PIONEER®

The Art of Entertainment

TUNING FORK

Audio and Video Service Guide No.10

Basic technology of Amplifiers

Part # : GRT1002

Table of Contents

Audio Amplifiers

1. Functions and Classifications

(1) Types of Amplifiers and Their Construction and Functions

An audio amplifier is basically configured as shown in Figure 1. The two main components are the preamplifier unit and the power amplifier unit. The three types of amplifiers generally available in the market are as follows:

① Preamplifier
② Power amplifier
③ Integrated amplifier

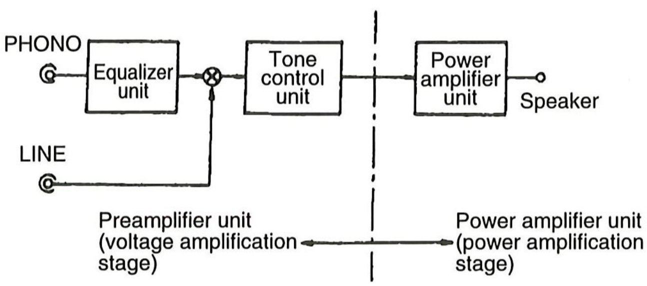


Figure 1 Audio Amplifier Configuration

To handle the various new types of audio and video components such as compact disc players and laser disc players appearing in recent years, it is becoming common for preamplifiers and integrated amplifiers to incorporate circuits for interfacing with and controlling digital and video signals, and Pioneer is placing much emphasis on developing products promoting these features.

In addition to basic amplifiers, various types of peripheral components are also available. These peripheral components are described in section 5.

Since signals output from source components such as compact disc players, record players, tuners, cassette decks and DAT decks are of too low a level to be used as is, a certain amount of pre-amplification is necessary. This and other functions such as input signal selection and tone (bass, treble) control are performed by the preamplifier.

As a high-power signal is required for producing large volume levels from speakers, the power level of the signal output from the preamplifier must be amplified and supplied to the speakers. This is the role of the power amplifier.

i. Preamplifier

A preamplifier generally has the following functions:

① Input signal selector
② Record player equalizer
③ Sound volume adjustment
④ Appropriate amplification of low-level input signals
⑤ Tone adjustment, filtering

A basic preamplifier configuration is shown in Figure 2 (a).

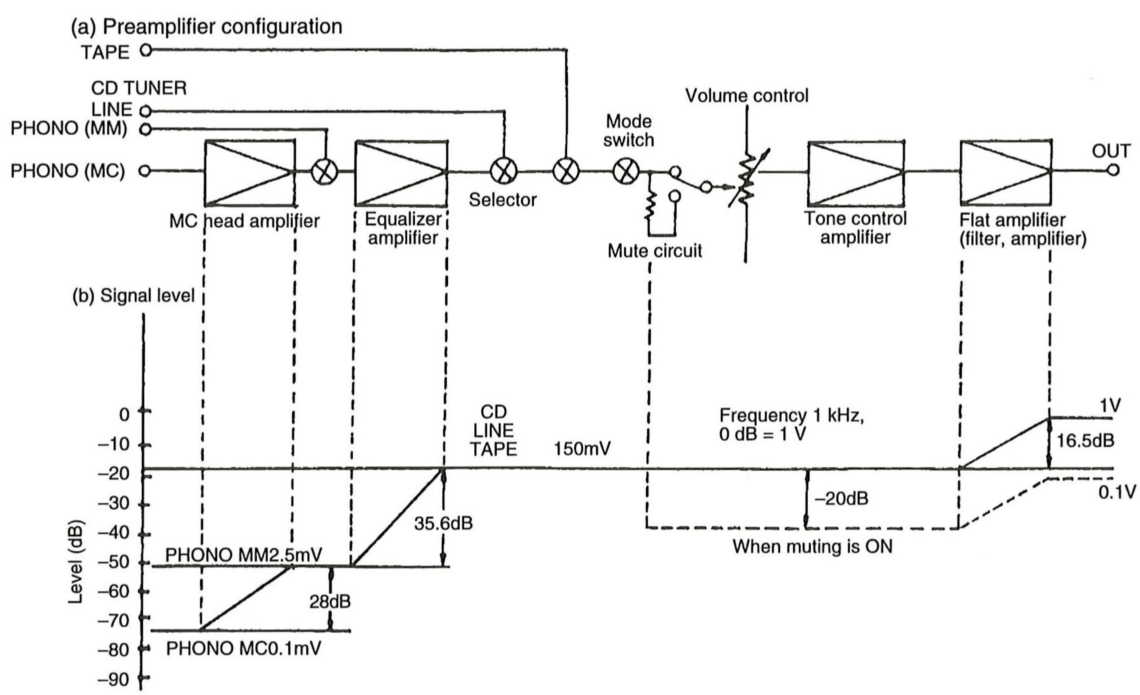


Figure 2 Preamplifier Configuration and Signal Level

The equalizer amplifier in Figure 2 is used to amplify signals input from analog records. When an analog record is made, the signals are recorded using RIAA recording characteristics which suppress the amplitude of low frequencies, so that the signal-to-noise ratio of high frequencies is improved. During playback, the equalizer amplifier uses RIAA playback characteristics which are the opposite of RIAA recording characteristics to return the signal to its original frequency characteristics.

When using an MC cartridge, the equalizer amplifier alone cannot provide sufficient amplification, so an MC head amplifier or MC transformer is used to preamplify the input signal before it enters the equalizer amplifier. In the MC transformer, the signal voltage is increased between the primary and secondary.

The tone control amplifier is a circuit used for tone adjustment, and is provided to allow compensation of the acoustics to match the speaker characteristics or speaker positioning, or to intentionally change the frequency characteristics to obtain a desired sound quality. The tone control allows independent adjustment of high and low frequencies. The device used to divide the frequency range into several divisions for fine tone adjustment is called a graphic equalizer.

The selector function used to switch between the various input signals and the volume control function used to adjust the input signal level are basic functions which are of absolute necessity for preamplifier operation. Other switches often found on preamplifiers such as mute, filter and loudness switches are auxiliary functions.

Figure 2 (b) shows the standard signal levels for each preamplifier circuit. The input sensitivity levels used by Pioneer are 2.5mV for PHONO MM signals, 0.05mV~0.3mV for PHONO MC signals, and 150mV for LINE signals from components such as CD players, tuners and tape decks. The preamplifier's output level is set at 1V.

ii. Power Amplifier

The role of the power amplifier is to take the signal which has been partially amplified by the preamplifier and boost the power of the signal to a level large enough to drive the speakers.

While the power amplifier operates using large signal levels (in terms of both voltage and current), it must also provide a good signal-to-noise ratio, wide-range frequency response and high operation stability. Also, since the output power is relatively large, the power amplifier must be designed with sufficient heat-dissipating characteristics to offset the heat generated during operation and prevent damage to the output transistors.

The ability of a power amplifier to drive speakers is indicated by its output power rating, which is normally in the 50W~200W range for general-use power amplifiers. Power amplifiers with large output power ratings of 500W~1kW have also been appearing in greater numbers recently. Most power amplifiers have an input sensitivity of around 1V.

Power amplifiers are generally marketed as either monaural amplifiers or stereo amplifiers, which are further classified into A class amplifiers and B class amplifiers depending on the operation state of the output stage.

iii. Integrated Amplifier

An integrated amplifier, as its name implies, combines a preamplifier and a power amplifier in a single unit. As shown in Figure 3, most modern integrated amplifiers have no pre out (preamplifier output) jacks and utilize a high-gain power amplifier unit, eliminating the flat amplifier. Also, it is common nowadays for integrated amplifiers to use a high-gain type equalizer amplifier instead of using an MC amplifier or MC transformer.

Integrated amplifiers are generally designed with multiple functions, while at the same time allowing the user to simplify the signal path by providing a "direct" switch for supplying the input signal directly to the power amplifier unit, bypassing the tone control circuitry.

By using these various methods to simplify the circuitry, audio makers can produce integrated amplifiers which provide higher performance, improved sound quality and multiple functions, as well as introduce advanced technologies while maintaining high cost performance.

(2) Signal Level Relationships with Other Components

An audio amplifier is connected with various other components such as CD players, tuners, DAT decks and cassette decks. If the signal levels and impedance values between the amplifier and these signal source components differed from component to component, level differences would occur between components causing

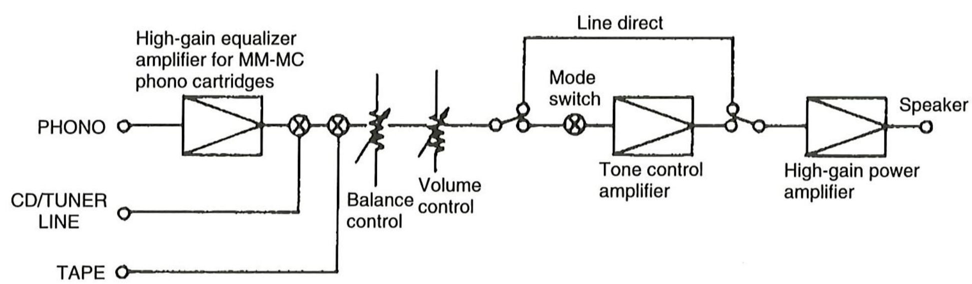


Figure 3 Integrated Amplifier Configuration

degradation in sound quality. To prevent this type of problem, signal levels and impedance values for amplifier input and output terminals must conform to certain standards (CP-102A) set by the EIAJ (**Electronic Industries Association of Japan**). Signal level and impedance standards for the various amplifier inputs are listed in Table 1, output level standards in Table 2, signal source component output level standards in Table 3, and input level and impedance standards for signal source components with input terminals in Table 4. The figures in these tables are taken from the CP-102A standards document. Also, as these standards were determined in 1979, values for CD players and DAT decks are not provided.

(3) Dynamic Range of Signal Source Components
The dynamic ranges of signal source components commonly connected to an audio amplifier are shown in Figure 4. The dynamic ranges of components constantly change as new technologies are developed. The ranges shown in Figure 4 are typical of high-grade separate components currently available in the market. In the figure, PHONO (LP) indicates the output level of the equalizer amplifier and AMP LINE IN indicates the level of the amplifier's LINE input. For general listening, it is common for the volume control of the amplifier to hold the input down to about -30dB. For an amplifier to be of practical use, therefore, the noise level must kept sufficiently below that point.

Table 1 Amplifier Input Terminals

Input terminals	Reference input level	Specified input level	Input impedance	Remarks
PHONO MC	0.21mV	0.15~0.3mV	47Ω~1kΩ	There are some actual amplifiers with input level ratings of approx. 0.1mV or less for use with **Ortofon** MC cartridges.
PHONO MM	2.1mV	1.5~3mV	47kΩ~100kΩ	2.5mV is used for most equipment.
MIC	4.25mV	3~8mV	47kΩ~51kΩ	
CD, TUNER, LINE, TAPE	142mV	100~200mV	more than 47kΩ	150mV is used for most equipment.
MAIN IN	—	355~1,000mV	more than 47kΩ	

Table 2 Amplifier Output Terminals

Output terminals	Conditions	Specified output level	Load impedance	Remarks
TAPE REC		100~284mV	47kΩ	
PRE OUT	Integrated amplifier	500~1,420mV	47kΩ	
	Preamplifier	710~2,000mV	22kΩ	
HEADPHONES	Speaker out provided	0.5~1.5mV	8Ω	With speaker of 1W/8Ω
	Speaker out not provided	0.1~0.4mV	8Ω	With specified input
The output impedance of amplifier output terminals is not particularly stipulated. From the standpoint of achieving optimum performance, however, it is generally the rule in audio systems to have low impedance outputs and high impedance inputs.				

Table 3 Signal Source Component Output Terminals

Output terminals		Specified output level	Load impedance	Conditions and Remarks
Record player	MC	0.15~0.42mV	470Ω	When playing reference records recorded with a velocity amplitude of 50mm/s (horizontal, monaural) or 35.4mm/s (45°, stereo).
	MM	1.5~4.2mV	47kΩ	
TUNER AM		100~284mV	22kΩ	Modulation: 30%
TUNER FM		100~284mV	22kΩ	Modulation: 30% (22.5kHz)
TAPE PLAY		150~300mV	22kΩ	When playing a cassette tape recorded at -4.5VU of 0VU (160nWb/m).
ADAPTOR OUT		100~284mV (500~1,420mV)	22kΩ	Values in parentheses ( ) are for PREOUT MAIN IN use.
The output impedance of signal source component output terminals is not particularly stipulated. From the standpoint of achieving optimum performance, however, it is generally the rule in audio systems to have low impedance outputs and high impedance inputs. Values for recent new types of components such as CD players and DAT decks are not stipulated. However, components which transfer and record digital signals from these components conform to the LINE system of the audio amplifier, and the maximum output level (at 0dB) of CD players and DAT decks is specified at 2V on the component side. Also, the average level at that point is approx. -20dB.				

Table 4 Signal Source Component Input Terminals

Input terminal	Reference input level	Specified input level	Input impedance	
TAPE REC, LINE IN		100mV/0VU	more than 47kΩ	0VU recording should be possible at 100mV.
ADAPTOR IN	142mV	100~200mV (500~1,000mV)	more than 47kΩ	Values in parentheses ( ) are for PREOUT MAIN IN use.

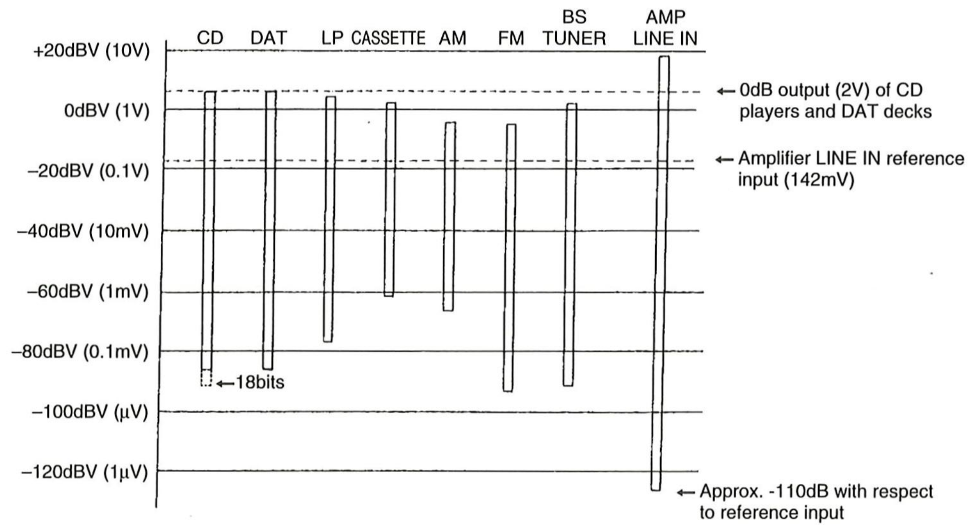


Figure 4 Dynamic Range of Various Signal Sources

2. Circuit Construction

As described in the previous section, audio amplifiers consist of a preamplifier and a power amplifier, with the preamplifier comprised of circuits such as an equalizer amplifier, MC head amplifier, tone controls and filter circuits. The tone control section is also sometimes referred to as an intermediate amplifier, and the power amplifier is also sometimes called the main amplifier. This section provides detailed descriptions of each circuit used in preamplifiers and power amplifiers, and the operation and characteristics those circuits supported by actual circuit examples.

(1) Phono Equalizer Amplifier

Analog records are recorded according to frequency characteristics determined by the RIAA (**Record Industry Association of America**). The amplitude vs. frequency characteristics used when cutting records is shown in Figure 5 (a). When the cut grooves are traced by a velocity sensing cartridge (MM, MC cartridge), the output frequency characteristics appear as shown in Figure 5 (b).

The equalizer amplifier has RIAA playback characteristics designed to compensate and flatten the cartridge output as shown in Figure 5 (c), and amplify the output to the same level (150mV) as CD and line (AUX) inputs. As shown in Figure 6, there are three types of phono equalizer amplifiers—NFB type, CR type and a hybrid CR-NFB type.

The NFB type is superior in terms of dynamic range and noise suppression. However, the capacitors used for the NFB element place a heavy load on the circuit at high frequencies, which can cause a rapid increase in distortion in the high range, reduce the through rate*1 and generate TIM distortion*2 if the circuitry is not well designed. Also, at ultra-high frequencies, deviations occur and the output does not drop below 0dB. Moreover, since a large current flows at high frequencies, separation and sound quality can easily be adversely affected if the cartridge is not properly grounded. However, as this circuit has a relatively simple construction, it is the most commonly used.

The CR type is the opposite of the NFB type, with dynamic range and noise suppression sacrificed to a certain extent for benefits in terms of sound quality.

The CR-NFB type combines the features of the CR and NFB types. Although the circuitry is somewhat complex, this design is frequently used for equalizers in medium to high class amplifiers. The CR-NFB type equalizer amplifier shown in Figure 7 performs turnover (T_H: amplification of low frequencies) with the initial stage amplifier, carries out roll-off (T_H: attenuation of high frequencies) with the next stage, and makes use of the overall DC negative feedback with the circuit to obtain low-range amplification stop characteristics (T_L). This design prevents degradation of the signal-to-noise ratio and dynamic range to a great extent and also improves DC stability (patent pending). In tests, equalizer deviation remains within ±0.1dB between 20Hz and 100kHz, and degradation of the signal-to-noise ratio is kept down to within 2dB of the NFB type, providing excellent audio performance.

An actual example of a CR-NFB type equalizer amp is shown in Figure 8. In this circuit, the final stage amplifier incorporates a subsonic filter (to be discussed in a following section), which effectively cuts ultra-low-frequency noise generated from record defects such as

*1 Through rate: The ability of the circuit to increase or decrease the voltage per unit time, expressed in "V/μs" units.

*2 TIM distortion: A type of transient distortion generated when the amplifier undergoes instantaneous saturation.

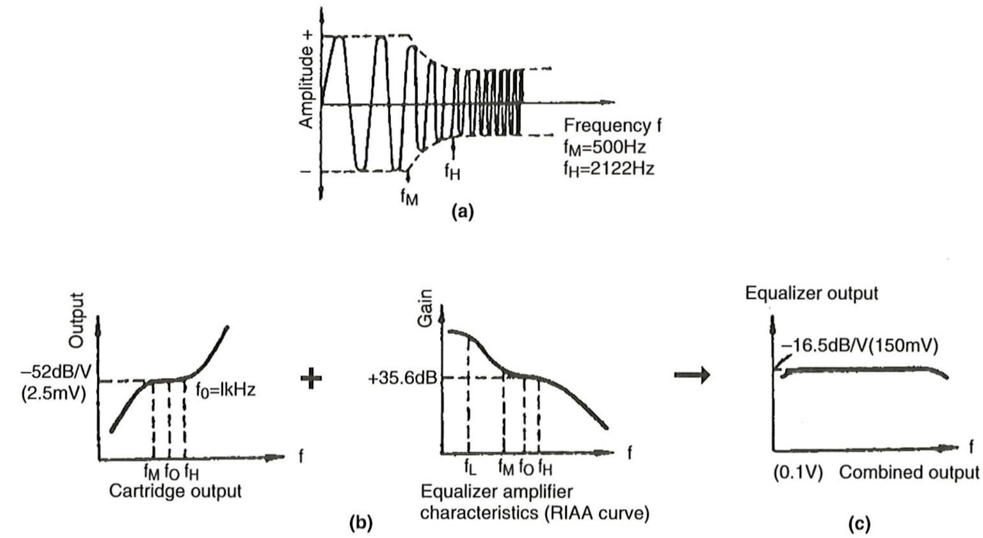


Figure 5 Equalizer Operation

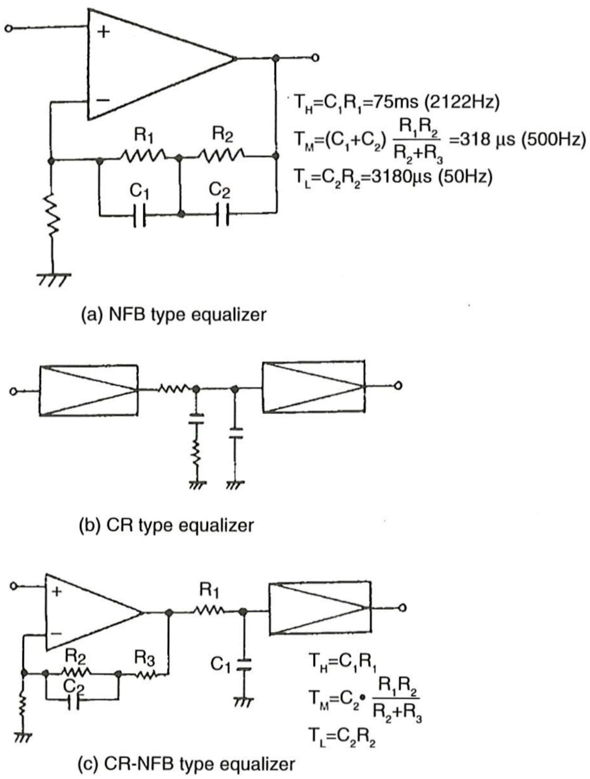


Figure 6 Types of Equalizer Amplifiers

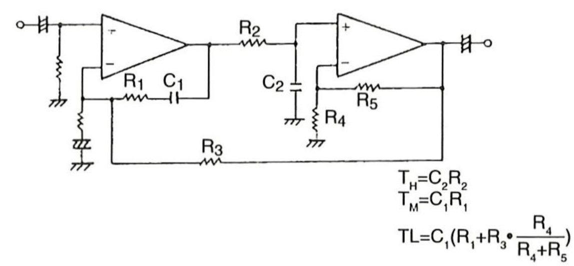


Figure 7 Example of CR-NFB Type Equalizer - 1

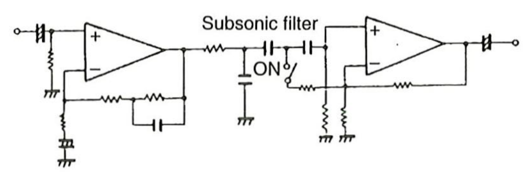


Figure 8 Example of CR-NFB Type Equalizer - 2

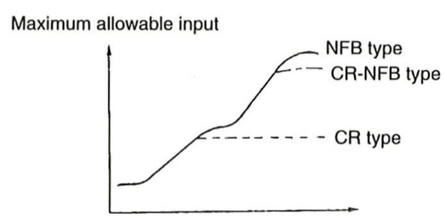


Figure 9 Dynamic Range Characteristics of Each Equalizer Type

warping.

It should also be noted that if roll-off is carried out at input section of the first stage of a CR-NFB type equalizer, the dynamic range will be improved but the impedance of the connected cartridge may cause the roll-off characteristics to change, and the series resistance may cause degradation of the signal-to-noise ratio. Furthermore, carrying out turnover in the final stage is undesirable in terms of low-frequency noise. Figure 9 shows the dynamic range characteristics of each circuit type.

(2) MC Head Amplifier

The output of an MC cartridge is extremely small, normally at a level of around 0.15mV at specified velocity amplitude (5 cm/s). This is close to the output level of a microphone or tape playback head, and requires amplification to bring it up to the equalizer amplifier's specified input level (2.5 mV). This is the task of the MC head amplifier—an amplifier which must generate very little noise and have very flat frequency characteristics.

Various types of MC head amplifier circuits are shown in Figure 10. Circuit (a) is a simple common base type

circuit. Circuit (b) is designed with several series-connected transistors in the initial stage to reduce thermal noise* (theoretically, by a factor of $1/\sqrt{n}$). The gain must be sufficient to amplify the signal from 0.15mV to 2.5mV, or by approximately 25dB. It is sufficient if an output can be obtained up to the equalizer's maximum allowable input, with an output level of 2~3V in the high-frequency range. Circuit (c) dispenses with the head amp and instead uses a high-gain design for the equalizer itself. The circuit is adapted for MC cartridge use by switching the gain. With this type of circuit, however, degradation of the signal-to-noise ratio is always worse than with circuits which use a dedicated MC head amplifier. Because of its simplicity, however, this type of circuit is widely used in low to mid-class amplifiers.
From the standpoint of thermal noise reduction, it is common for designers to use a circuit such as shown in Figure 10 (d), which employs a step-up transformer instead of an amplifier. Although use of a step-up transformer is generally inferior to an amplifier because of its narrower band and greater induction hum and distortion, a good design can both reduce thermal noise and provide a good signal-to-noise ratio. For this reason, this type of circuit has started to regain popularity in recent years. The hybrid type amplifier, described later in section 4, uses both an amplifier and a transformer, effectively taking advantage of the good points of both designs.

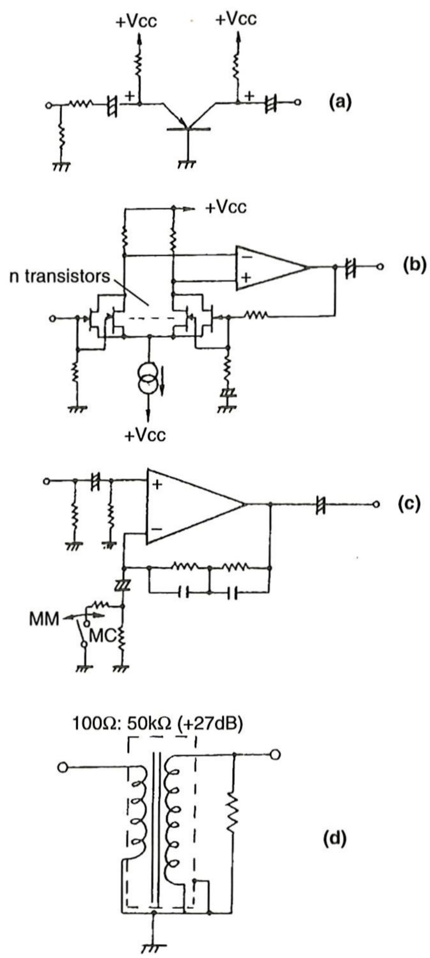


Figure 10 MC Head Amplifier Circuits

(3) Flat Amplifier

The flat amplifier, having flat frequency characteristics, is used for supplying additional gain in circuits where the gain from other amplifiers is insufficient. Flat amplifiers are also used as buffer amplifiers and mixing amplifiers. A flat amplifier has smaller gain than an equalizer amplifier, but it must provide the same level of dynamic range, distortion and signal-to-noise characteristics.
The circuit shown in Figure 11 (a) is a commonly used flat amplifier. Caution is required, however, because of C_f is inserted with the objective of attenuating the high range, the level will not attenuate below 0dB. Also, if a long output line is led from an amplifier containing a large amount of negative feedback, oscillation and unstable operation is likely. This type of instability can be effectively prevented by inserting a several hundred ohm resistor (R_0) in the output line.
Figure 11 (b) shows the case of a flat amplifier used as an inverting type mixing amplifier. The gain for each input is virtually equivalent to R_f/R_s.

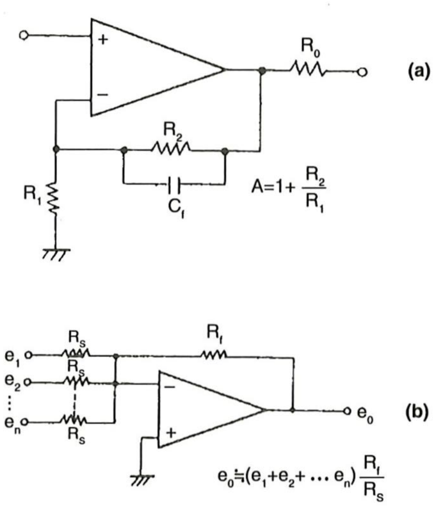


Figure 11 Flat Amplifiers

* Thermal noise: Also called Schottky noise
$e_n^2 = 4kTR\Delta f$

e_n^2	: Thermal noise mean square value
k	: Boltzmann's constant (1.38×10^{-23} [J/°K])
R	: Resistance value within the specified band
Δf	: Frequency bandwidth
T	: Absolute temperature

(4) Tone Control

The tone control circuit adjusts the quality of the sound by varying the low and high frequency levels. The two main types of tone control circuits are the general type which adjusts tone along a 6dB/octave gradient and the graphic equalizer type. Details of the graphic equalizer type are provided in section 5, Peripheral Components.
The tone control circuit is provided to allow adjustment of the sound quality by compensating for excess or insufficient high-frequency sound. The adjustment range is generally around ±8~10dB at center frequencies of 100Hz and 10kHz. The types of general tone control circuits in use today are the CR attenuating type, the series feedback type, and the BAX and LUX parallel feedback types, as shown in Figure 12. The LUX type circuit is a variation of the BAX type, adding a single capacitor to the low-frequency adjustment section to eliminate the "waviness" characteristic of the BAX circuit, resulting in superior tone variation characteristics. Design of these parallel feedback circuits is relatively simple, but because a large R_S is always in series with the signal, signal-to-noise ratio degradation is generally worse than with series feedback circuits. A B-curve potentiometer is used for the variable resistor.
The CR attenuating type circuit does not necessarily require an amplifier if the signal source resistance is low enough. Because of this, this type of tone control circuit was commonly used in popular-class amplifiers in the past. However, since the circuit's insertion loss is nearly 20dB, further amplification must be added somewhere to compensate for the loss, resulting in overall degradation of dynamic range. Because of this, the CR attenuating type circuit is rarely used at present. The variable resistor for the CR attenuating type circuit is a potentiometer with A-curve characteristics.
The series feedback type circuit (d) is widely used in many different forms. The series feedback circuit shown in the figure is constructed by simply inserting the CR type circuit into a feedback circuit, with the boost/cut orientation of the potentiometer opposite of the CR type circuit. Moreover, a potentiometer with C-curve characteristics is used. When using potentiometer with a 20C curve (opposing rotation angle resistance variation characteristics with the resistance value at the center position equivalent to 80% of the total resistance) set at the flat (center) position, gain of approximately 4X (12dB) is obtained with the input and output equal in phase. This series feedback circuit is widely used in popular-class amplifiers today because a tone control function can be added while maintaining a simple circuit construction by simply inserting the circuit into the feedback circuit of a power amplifier (power amplifier NF tone). However, since the amount of feedback increases significantly when the high range is attenuated, consideration is necessary to prevent the amplifier from becoming unstable.

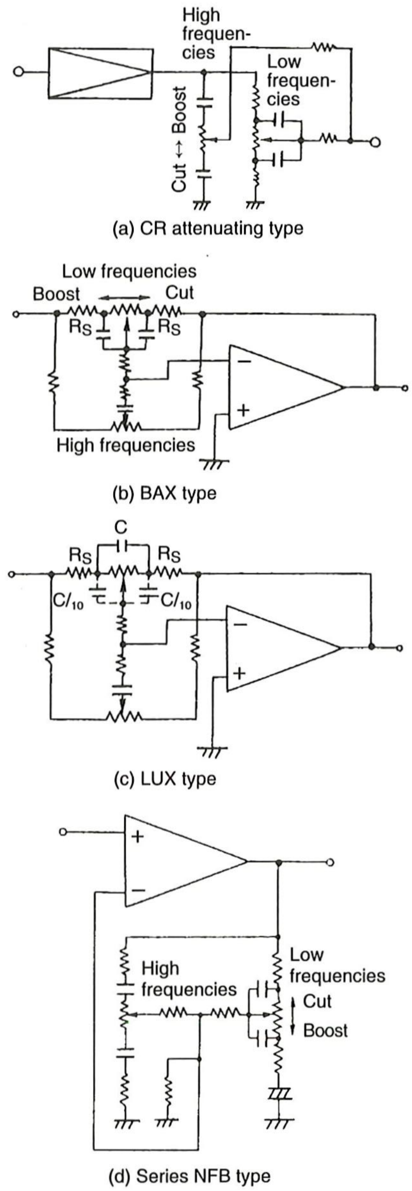


Figure 12 Tone Control Circuits

(5) Accessory Circuits

Circuits which perform supplementary functions other than the main amplifier functions are called accessory circuits. Accessory circuits include filters, loudness control circuits and "new function" circuits.

i. Filters

Filters used in amplifiers include low-cut filters for eliminating low-frequency noise such as the hum of the motor used when playing a record or tape, wow and flutter noise and ultra-low-frequency noise from record warps, and high-cut filters for cutting high-frequency noise such as record scratch noise, type hiss and tuner multiplex (MPX) noise. Cut-off frequencies are generally 7Hz~100Hz for low-cut filters and 7kHz~15kHz for high-cut filters, although filters which cut into the audible frequency range cause noticeable degradation of

playback sound. Therefore, cut-off frequencies should be set outside the audible range (below 30Hz for high-pass filters and above 15kHz for low-pass filters). Low-frequency cut-off filters are also frequently called subsonic filters.
It is desirable for the attenuation characteristics of these filters to have a steep slope (6 dB/octave to 12 dB/octave) outside the band, with feedback type filters generally exhibiting better performance than CR type filters.
Also, for maximum sound quality, it is desirable that the filter provide extremely uniform group delay characteristics*1 so as not to disrupt frequency characteristics within the band.
The filters in Figure 13 are widely-used 12 dB/octave feedback type low-pass and high-pass filters. One point of caution when constructing a high-pass filter is that if the emitter current is set fairly low and a large output impedance is present, it will become impossible to sufficiently attenuate high frequencies. This occurs because at high frequencies the input signal voltage is divided between R_1 and the output impedance. To prevent this from happening and to hold distortion to a low level, it is necessary for the emitter current to be set sufficiently large.

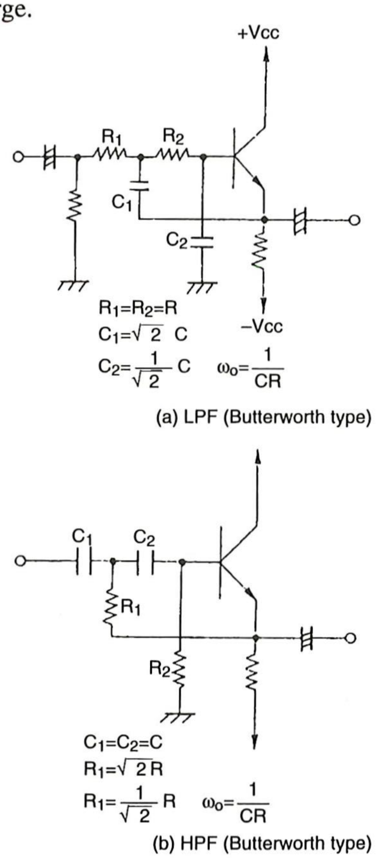


Figure 13 12 dB/octave Filter Circuits

ii. Loudness Control
The loudness control circuit shown in Figure 14 (a) compensates audible frequencies according to the Fletcher-Manson curves*2 shown in Figure 14 (b). This circuit is linked to a potentiometer to control the frequency response so that high and low frequencies are boosted at low sound volumes, with the effect flattening out at higher sound volumes. The effect of this circuit makes the playback sound appear flat to the human ear even when listening at low volume levels.
Also on the market are dynamic loudness circuits which carry out finer control of the frequency response by detecting the signal level and adjusting the frequency characteristics based on that level. For even more precise compensation, it would also be possible to create a dynamic loudness circuit which would predict the sound pressure level reaching the human ear.

iii. Bass Synthesizer
Many movie sources such asLaserDiscs and videotapes have an extremely compressed low-frequency range, and it is also common for the playback frequency response of speakers to drop sharply near the low-frequency end of the sound reproduction range. By generating a half frequency component of the low-frequency signal within

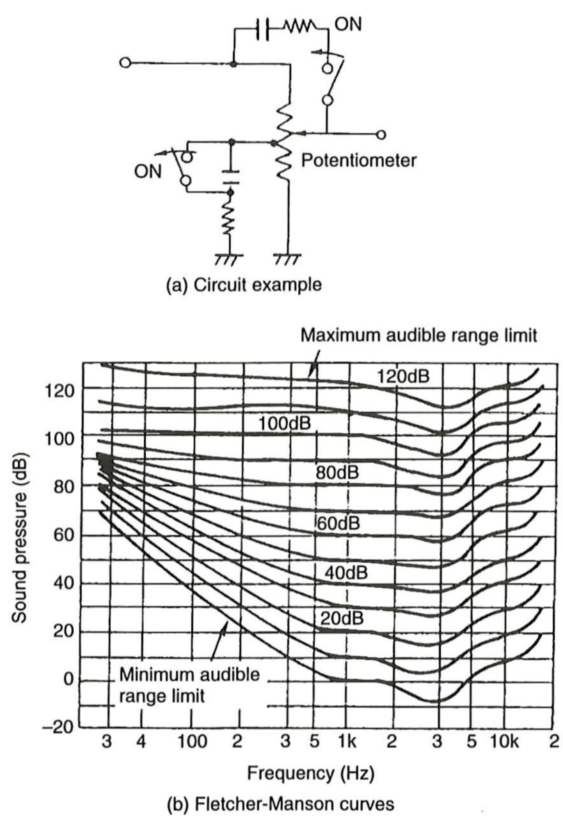


Figure 14 Loudness Control

*1 Group delay: Generally expressed by the symbol τ in [s] units.
$\tau = d\phi/d\omega$
"Uniform τ characteristics" means that the same delay is applied to all frequencies, thus preserving the shape of the original transmission waveform.
*2 Fletcher-Manson curves: Human ear hearing sensitivity characteristics graphed with respect to sound pressure level.

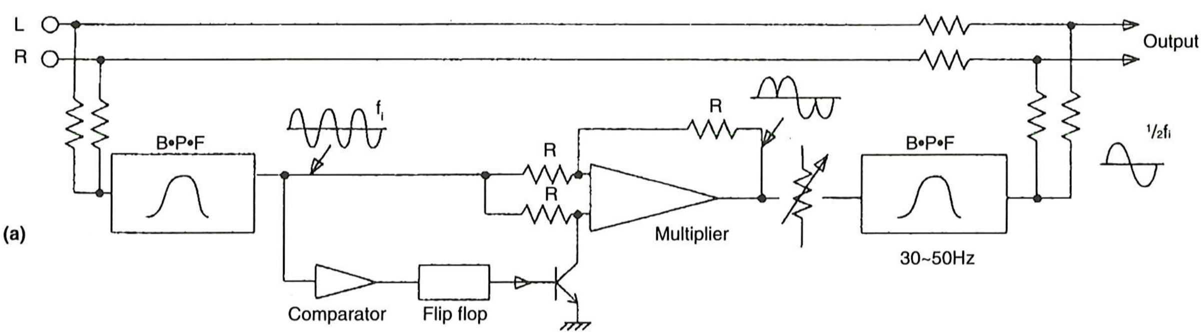
the amplifier and adding it to the source sound, the lower end of the frequency range appears to the listener to expand by about an octave, reproducing a sound field that is more powerful and "live" sounding than was previously possible. However, due to the increased load on the speakers as the playback range is extended to lower frequencies, the low frequency limit must be set at around 20~30Hz. In other words, it is generally good practice to set the input bandpass filter (BPF) cut-off point at 40~60Hz or greater.

A very simple synthesizer circuit is shown in Figure 15 (a). Ideally, however, it is desirable to divide the input BPF into several parts as shown in Figure 15 (b) and recombine the signal at the output side. The trick to designing a natural-sounding synthesizer is to set a dead zone of around 50mV in the comparator to prevent boomy bass sound and misoperation caused by noise and dialogue (voice).

Compared to the VCA (voltage controlled amplifier) method which generates a source signal envelope time delay, this bass synthesizer circuit differs in that the amplitude information of the source signal is maintained in real time, realizing a natural and fast-feeling synthesizer which does not sound abnormal even to musicians and audiophiles.

iv. Stereo Wide

In conjunction with development of the movie culture and increasing popularity of "surround" sound field reproduction in recent years, there is a strong demand for sound field reproduction providing a "live" atmosphere. The stereo wide circuit is a circuit that has a simple construction and can provide an expansive, surround-type sound field without the use of rear speakers. Furthermore, a more effective sound field can be obtained by adding rear speakers as a speaker matrix. Because of these features, this circuit is widely used in popular-class integrated amplifiers and in the amplifier units of combined component products. However, the properties of the circuit require caution during use because (1) no effect is obtained with monaural signals, (2) the effect differs according to the recording source, and (3) undesirable sounds from sources with much distortion or noise (such as a record player with a mistracking cartridge) are emphasized. For monaural signals, the simulated stereo circuit described in the following section can be used in conjunction with the stereo wide circuit to good effect. An example of a stereo wide circuit is shown in Figure 16. In this circuit, C_S is provided to obtain a greater effect at frequencies above that of the human voice*, and C_f is provided to reduce



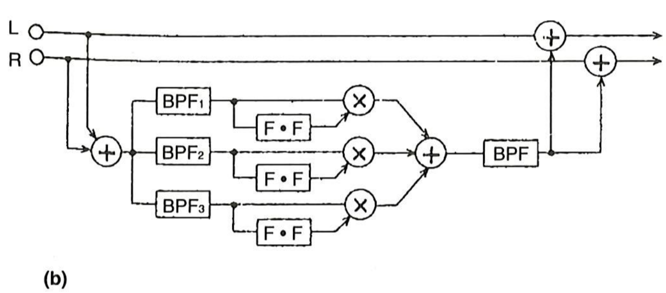
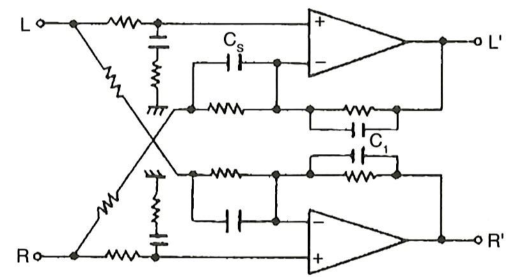


Figure 15 Bass Synthesizer Circuit



$\begin{cases} L' = |L|_{LPF} + K(L-R) \\ R' = |R|_{LPF} + K(R-L) \end{cases}$ K: Constant or transmission characteristics | $|_{LPF}$ is the signal passing through the LPF

Figure 16 Stereo Wide Circuit

* If an attempt is made to apply the effect to the vocal band, the sound image position will become vague and indistinct.

the adverse effects of high-frequency noise present in the signal.

v. Simulated Stereo

There are three basic methods of achieving a quasi-stereo effect with monaural signals from sources such as videos and AM radio broadcasts:

① The phase-shift method whereby a phase difference is placed between the left and right signals;

② Achieving a feeling of directionality by inserting a delay between the left and right signals;

③ Simulating a stereo signal by using a comb filter to divide the signal into narrow bands and shifting the bands of both channels in a complementary manner.

With any of these methods, excessive processing of the signal will result in an unnatural sound field. For a natural sound field, therefore, the phase shift should be kept within 90° when using the phase shift method, the time delay should be kept within 20~60ms when using the delay method, and the bandwidth divisions should not be smaller than several hundred Hz when using the comb filter method. The phase shift method is the most commonly used at present. The other two methods are not generally used because of their relatively high cost. Circuit examples and concept diagrams of these methods are shown in Figure 17.

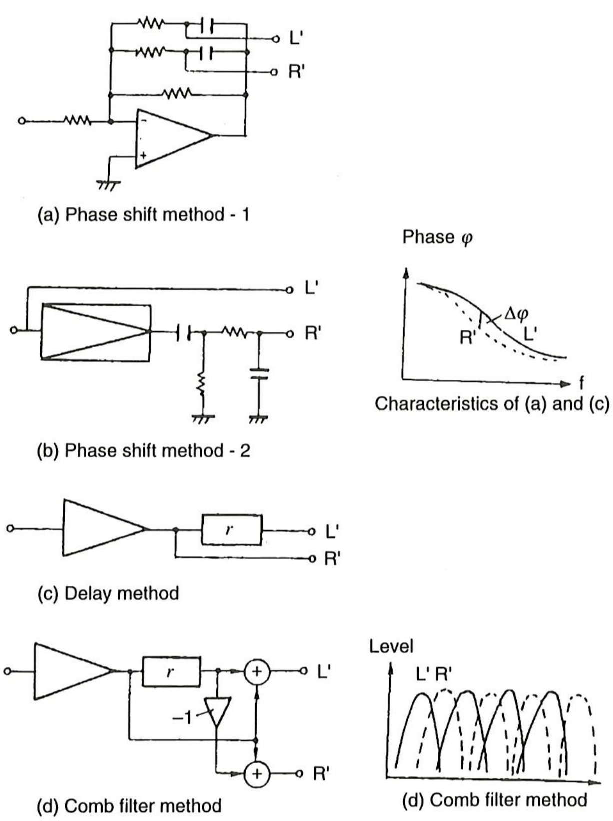


Figure 17 Simulated Stereo Circuits

(6) Power Amplifier

A power amplifier is the primary amplifier used to drive the speakers, and is thus often called the main amplifier. The basic construction of a power amplifier can be broadly divided into four stages: the initial stage, the pre-driver stage, the driver stage and the output stage. In high-class amplifiers, there are also cases where second and third stages are inserted between the initial stage and the pre-driver stage to increase the gain, or where the pre-driver stage is divided into two stages. The initial to pre-driver stages are collectively called the voltage amplification stage, and the driver to output stages are collectively called the power amplification stage. Most power amplification stages employ an emitter-follower construction (common collector amplification), which in actuality amplifies current rather than power.

Various possible output stage configurations include SRPP (shunt regulated push pull), DEPP (double ended push pull), SEPP (single ended push pull) and BTL (balanced transformerless). However, due to the technology currently available which allows free combination (complementary connection) of transistors with different polarities (pnp, npn), emitter-follower type SEPP and BTL configurations are most common today. Amplifiers which have no coupling capacitor for the output are called OCL (output condenserless) amplifiers. The OCL configuration is used in most popular and higher class amplifiers with positive-negative (dual pole) power supplies. Amplifiers in radio cassette players and portable audio equipment powered by batteries or a positive (single pole) power supply, however, commonly have a capacitor in the output. There is a significant difference in performance depending on whether a capacitor is included in the output, with the difference especially noticeable in the areas of low-frequency output, frequency response, phase and output impedance.

The BTL configuration is designed for enabling an OCL design for amplifiers equipped with only a positive (single pole) power supply, and requires use of two power amplifiers of the same type. This type of configuration provides several benefits. For example, as long as the power supply voltages are the same, a theoretical power output four times the output of a single power amplifier is gained, and the fact that no speaker current flows in the ground line is beneficial in obtaining high sound quality. Due to these advantages, the BTL configuration is widely used even in amplifiers with positive (single pole) power supplies.

The various output stage configurations are shown in Figure 18. When using an OCL design with any of these configurations, careful attention must be paid to DC stability to minimize the DC component flowing to the speakers (generally to within ±100mV).

Recently, it is becoming common to use circuits combining differential amplifier circuits and current mirror circuits in the initial stage, applying 100% DC negative feedback to achieve an extremely stable (low drift) amplifier design. Also, since capacitors are completely eliminated from the signal path, it was popular for a while to use a DC servo design which detects the DC component of the output and feeds it back to the input. The principles of the differential amplifier and current mirror circuits are shown in Figure 19. These two circuits—which can also be called fundamental transistor amplifier circuits—are making significant contributions to recent amplifier technology developments and performance improvements.

Figure 20 (a) shows a typical power amplifier circuit used in popular-class amplifiers, consisting of a differential amplifier ($Q_{1,2}$) and current mirror (Q_3) in the initial stage, a pre-driver (Q_4), a driver ($Q_{6,7}$) and an output stage ($Q_{8,9}$). $R_1 : R_2 \fallingdotseq 1 : 3$ is used for two-stage power amplification, and $R_1 : R_2 \fallingdotseq 1 : 5$ is used for three-stage power amplification.

Figure 20 (b) shows a typical power amplifier circuit found in mid-class and high-class amplifiers. An FET replaces the input coupling capacitor in the initial stage, eliminating one cause of sound quality degradation.

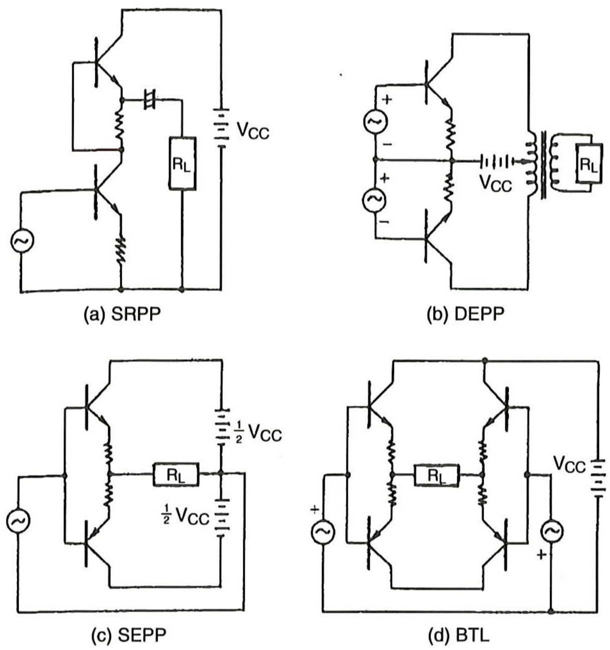


Figure 18 Various Output Stage Configurations

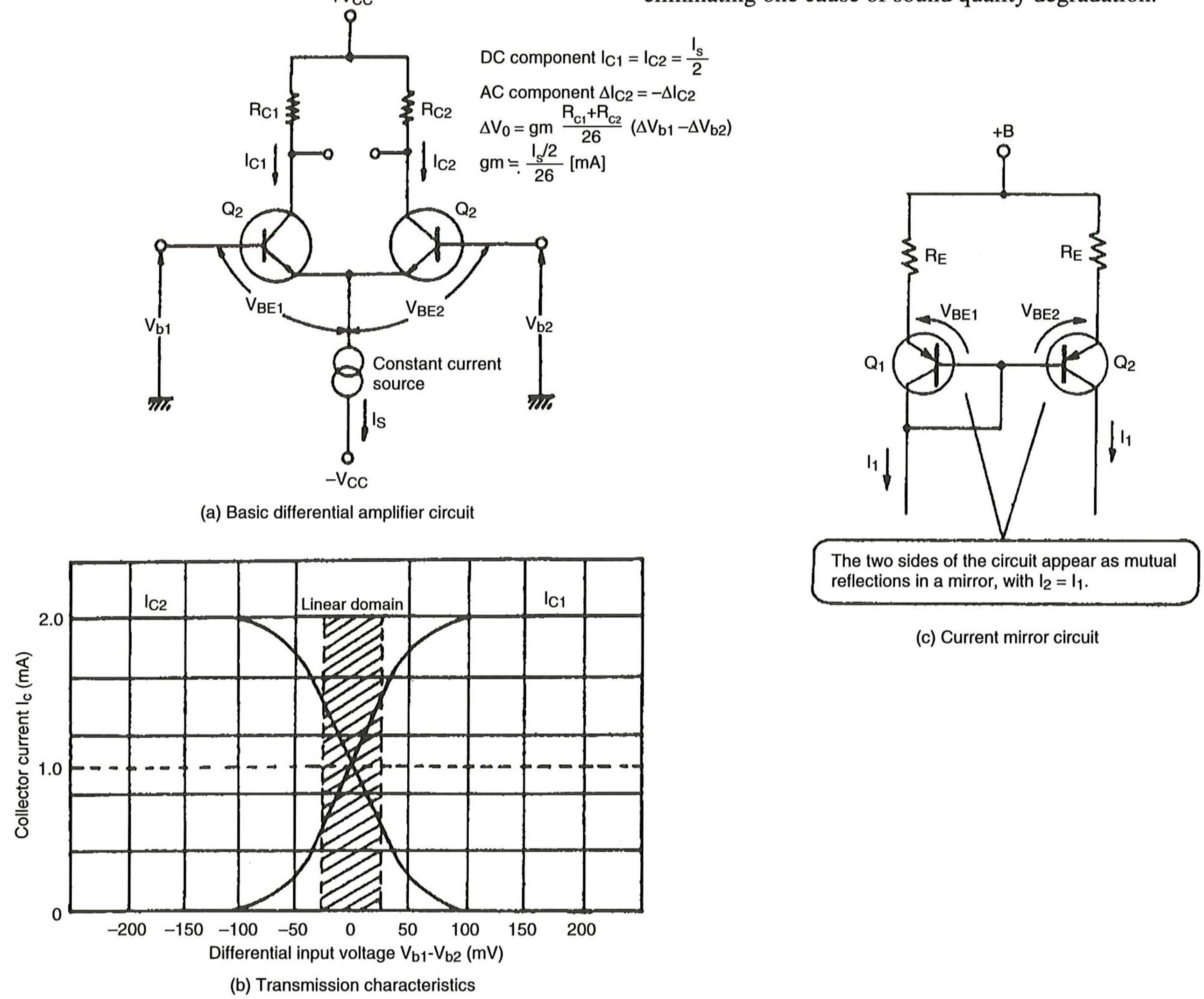


Figure 19 Basic Differential Amplifier and Current Mirror Circuits

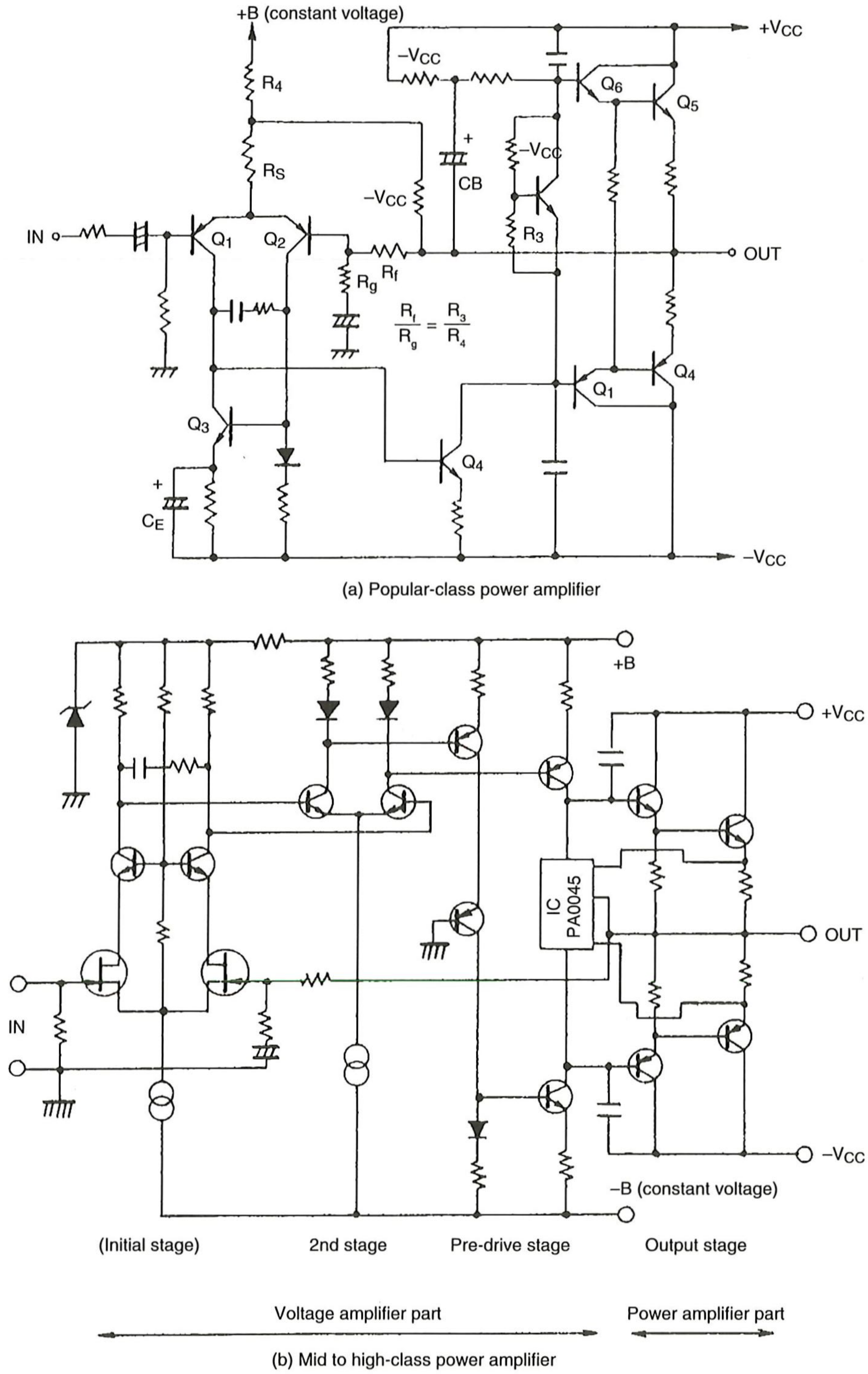


Figure 20 Typical Power Amplifier Circuit Examples

Also, the initial stage load—called a cascode bootstrap—cancels the equivalent amount of capacitance in the input, improving high-frequency response and eliminating distortion caused by unbalanced fluctuations in the amplification voltage (distortion based on the Early effect). PA0045 is a custom IC designed by Pioneer, and operates as an NSC (Non Switching Circuit) Type II (described in section 4). This eliminates the need for temperature compensation and helps to improve performance.

The above is a general overview of how power amplifier circuits are constructed. Although recent high-class amplifiers contain various complex circuit technologies developed independently by each company, all of them employ feedback techniques and a combination of differential amplifiers, current mirrors and cascode circuits.

(7) Power Supply Circuit

All the energy necessary for driving the speakers is supplied by the power supply. Thus, if a weak power supply is used, the amplifier will not be able to

sufficiently drive speakers which exhibit complex impedance fluctuations, resulting in degradation of sound quality. Although the basic performance required of an amplifier power supply circuit differs slightly depending on whether the amplifier is a preamplifier or a power amplifier, the basic requirements are as follows:

① The circuit should have low output impedance.
② The circuit should produce very little ripple or noise.
③ The circuit should have plenty of current capacity.
④ Protection measures should be built in to protect against malfunctions or abnormalities.

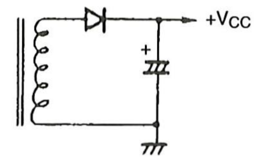
Also, when a high gain circuit such as a preamplifier coexists in the same unit as a power amplifier circuit, the layout must be carefully planned to prevent electrostatic and electromagnetic cross-interference. If sufficient attention is not paid to the layout, there is possibility of undesirable crosstalk, separation, sound quality degradation and even amplifier oscillation.

i. Rectification Methods

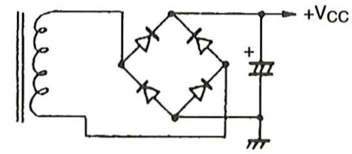
Various power supply rectification networks are shown in Figure 21. Bridge rectifiers are commonly used in power amplifier power supplies because of their relatively high transformer utilization rate. Full-wave rectifiers are used in preamplifier power supplies, and half-wave or voltage doubler rectifiers are commonly used in power supplies for circuits with small power consumption such as bias circuits. Regardless of which type of rectifier is used, it is important to use diodes and capacitors with strong surge resistance characteristics to absorb the effects of the large surge current that occurs when the power is turned on. Moreover, since the pulse-shaped rectified ripple current may be several times the size of the DC output current, capacitors must possess strong ripple resistance characteristics. For good ripple resistance, it is generally better to use capacitors with large rather than small capacitance, or to connect several capacitors of the same capacitance in parallel.
The rectifier network shown in Figure 21 (e) divides the transformer winding into two parts to allow independent positive and negative rectification. This design improves sound quality by eliminating power source interference between positive and negative poles, and is therefore recently being increasingly used in high-class amplifiers.

ii. Voltage Regulators

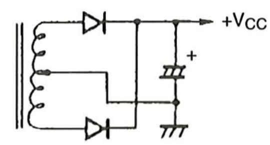
High gain circuits such as used in the initial stage of preamplifiers and power amplifiers are subject to oscillation (motorboating), signal-to-noise ratio reduction and sound quality degradation if power supply fluctuations, ripple noise and impedance are not kept to a minimum. After the rectification stage, therefore, it is common to insert a voltage regulation circuit to ensure a stable supply of power. Feedback type voltage regulator circuits are most common, although simple type circuits are also widely used in popular-class amplifiers.
Lower output impedance is possible with feedback type circuits, but ultra-high-frequency(VHF~UHF) oscillation may occur if the circuit is not designed well. Recently,



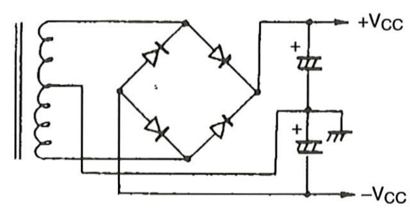
(a) Half-wave rectifier



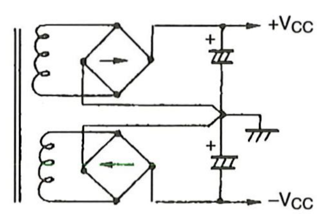
(b) Full-wave rectifier



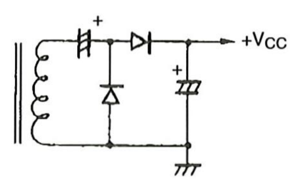
(c) Center tap full wave rectifier



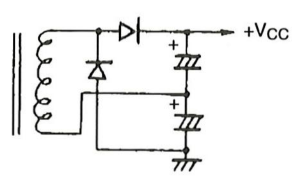
(d) Center tap bridge rectifier



(e) Positive-negative divided winding bridge rectifier



(f) Half-wave voltage doubler rectifier



(g) Full-wave voltage doubler rectifier

Figure 21 Rectification Methods

feedback type circuits are being implemented as monolithic ICs and used widely as three-terminal regulators. Examples of feedback type and simple type voltage regulator circuits are shown in Figure 22 (a) and (c). The circuits shown in Figure 22 (b) and (d) are called shunt regulators, because the signal current loop is terminated

within the circuit block. This design prevents the current from affecting other circuits, thus improving sound quality. However, since a large current is constantly flowing, power consumption is large.

If the objective is more to minimize ripple noise than to stabilize the voltage level, a ripple filter like that shown in Figure 22 (e) is used. With this circuit, the capacitance as seen from the emitter is multiplied by a factor of h_{FE} (resistance $R = 1/h_{FE}$), and the voltage loss is minimal (around 0.8V), making it an effective filter.

(8) Protection Circuit

If the amplifier speaker terminals are accidentally shorted while making connections, excessive current could damage the power transistors in the output stage, particularly with large output amplifiers. Or on the contrary, if a malfunction occurs in an OCL power amplifier, generating a large DC voltage in the output, the speakers may burn out.

A protection circuit is an accessory circuit incorporated in a power amplifier to prevent these types of occurrences which could cause damage. A protection circuit is also used for purposes such as providing a muting function for preventing the pop noise or distortion heard when the power is turned on and off. A protection circuit should only operate in abnormal circumstances like those described above or when an excessive operation occurs. It is therefore important to design the circuit so

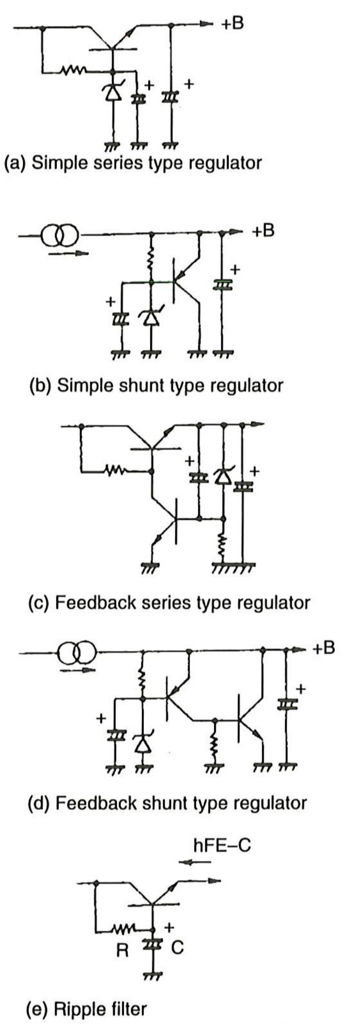


Figure 22 Voltage Regulators

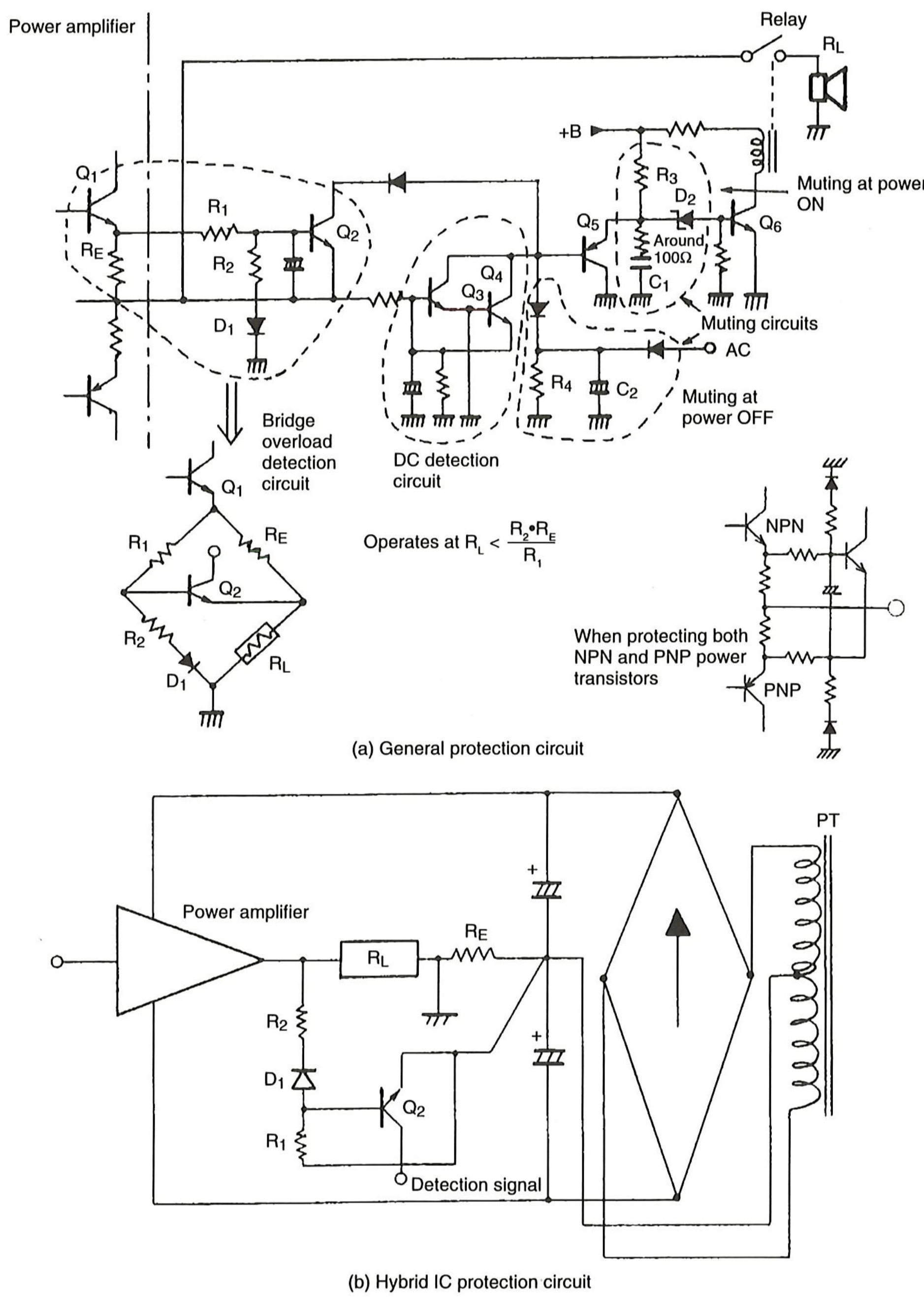


Figure 23 Protection Circuit Examples

that it does not accidentally operate at undesired times, and that even when the protection function activates, normal operation will be restored when the abnormal operation status is cleared.
Figure 23 (a) shows a protection circuit commonly used in Pioneer products. This circuit design combines the following functions in a single circuit:

- Bridge type overload detection function
- Mid-point DC voltage detection function
- Power ON/OFF muting function

i. Bridge Type Overload Detection

If an amplifier's output terminals are shorted or too many speakers are connected, causing an excessive reduction in load impedance, the power that must be handled by the output stage transistors becomes excessive, damaging the power transistors.
In such a case, the balanced state of the bridge detection section of the circuit in Figure 23 (a) crumbles and Q_2 turns on, which causes Q_5 to turn on and Q_6 to turn off, opening the relay contacts and disconnecting the speakers from the power amplifier.
Figure 23 (b) shows a bridge detection circuit used for protecting circuits such as hybrid ICs where overload cannot be detected from R_E.

ii. Mid-point DC Voltage Detection

If a malfunction occurs in a component such as a power amplifier's output transistor, direct current may be output through the output terminals. With OCL type circuits, this would cause direct current to flow to the speakers, resulting in damage to the speakers. With the circuit in Figure 23 (a), if a DC potential greater than a certain level is generated in the power amplifier output, Q_3 (positive detection) or Q_4 (negative detection) turns on, causing the relay contacts to open and disconnect the speakers from the power amplifier.

iii. Power ON/OFF Muting

When the amplifier power is turned on or off, the amplifier's DC operation becomes unstable for an instant, causing a "pop" sound or distortion to be heard from the speakers. This sound is prevented by a muting circuit which makes use of the protection circuit's relay to disconnect the speakers from the amplifier when the power is turned on and off.
Muting operation when the power is turned on is as follows: The time constant circuit formed by R_3 and C_1 [Fig. 23 (a)] delays the rise in base potential of Q_6. When the power is turned on, current flows through R_3 charging C_1 and increasing C_1's voltage. When this voltage exceeds the Zener voltage of D_2 (Zener diode), Q_6 turns on to close the relay contacts and connect the speakers to the amplifier.
When the power is turned off, C_2 rapidly discharges through R_4, turning Q_5 on and Q_6 off to quickly open the relay and disconnect the speakers.
Also, when the abnormal condition which activated the protection circuit is cleared, the sound is muted for a certain period before the unit resets.

3. Performance

This section explains the meanings and descriptions of terms used for indicating amplifier performance in the specifications section of brochures and catalogs, based on the measurement methods (CP301A) and indication methods (CP-107) specified by EIAJ (EIA).

(1) Rated Output

Power amplifier output rating methods can be broadly classified into two types: (1) output ratings such as effective power, continuous power and rated power indicating levels of power which can be continuously produced, and (2) output ratings such as music power and dynamic output indicating power levels which are only produced for an instant. According to EIAJ standards, continuously available power is specified as rated output or practical maximum output, and instantaneously produced power is specified as dynamic headroom.
In general, due to the power source circuitry and components used, power produced at high and low frequencies is less than at middle frequencies. Because of this, the output rating value will differ depending on whether measurements are made for only a frequency of 1 kHz or for the entire frequency range. Figure 24 shows a frequency vs. output curve which illustrates why the output rating for the 20Hz~20kHz band is lower than the output rating at 1 kHz. Thus, rated output can be expressed for either a specified frequency range or the single frequency of 1 kHz. At this time, other ratings such as distortion and load resistance which affect the

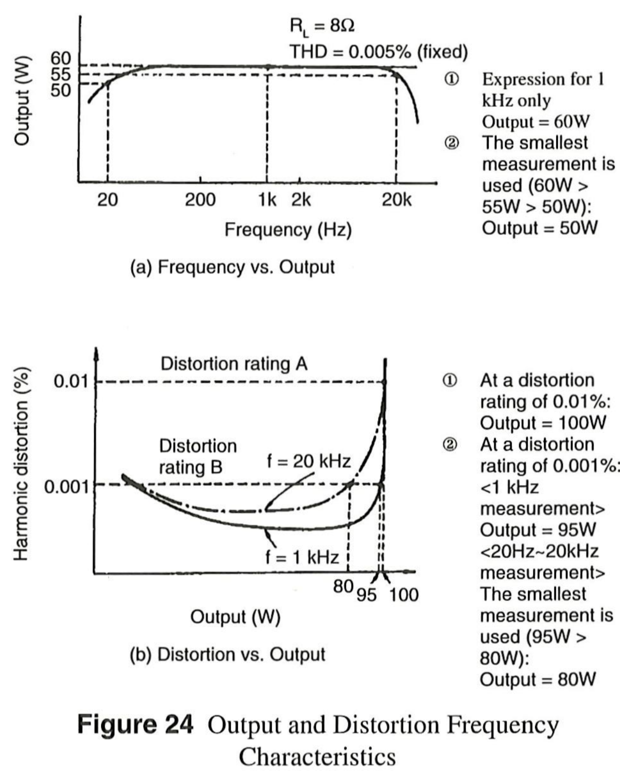


Figure 24 Output and Distortion Frequency Characteristics

output rating are specified by the manufacturer. Another way to express the output rating is as "practical maximum output," which is the power output measured at a frequency of 1 kHz and 10% distortion.

Examples of output rating expressions:

- Rated output 100W + 100W (20Hz~20kHz, 0.005%, 8Ω)
- Rated output 80W + 80W (1kHz, 0.005%, 8Ω)
- Practical maximum output 50W + 50W (EIAJ)

Output ratings using one or more of the above expression methods are included in the specifications listing.

(2) Rated Distortion

This is the distortion rating used when measuring the rated output just described, and can be specified independently for each model by the manufacturer.

(3) Total Harmonic Distortion

This is the distortion rate of the amplifier measured at a specified output at a single frequency of 1 kHz or over a specified frequency range. The specified output is usually 1/2 or 1/10 the rated output or a specific figure such as 1 W, and can be seen as a realistic output typical of actual use conditions.

(4) Dynamic Headroom and Dynamic Power

Dynamic headroom* is a ratio between instantaneous output and rated output, expressed as follows:

$$\text{Dynamic headroom (dB)} = \frac{\text{1 kHz burst wave maximum output (W)}}{\text{Rated output (W)}}$$

However, since it is difficult to get a feeling for the actual output from the dynamic headroom figure, most manufacturers list the 1 kHz burst wave maximum output rating by itself, calling it "dynamic power."

This 1 kHz burst wave maximum output rating is measured by inputting the 20ms burst signal wave shown in Figure 25, observing the output wave, measuring the greatest peak-to-peak voltage within the 20ms interval which does not cause clipping, and calculating the output power from that voltage.

Different views of this dynamic power rating have caused debate in recent years. While 20ms is not a very long time interval in terms of music, it is much more indicative of actual use conditions than the previously used "music power" rating (the maximum output obtained while maintaining the DC voltage generated with no input signal). Especially with small loads, dynamic power has recently attracted attention as an accurate measure of an amplifier's small-load drive capability. Moreover, since all audio equipment makers are now using the same dynamic power measurement standards, accurate comparisons can be made between amplifiers of different makes.

(5) Output Bandwidth

The output bandwidth is the range of frequencies at which the amplifier can supply half its rated output into a load at a specified rate of distortion. The output bandwidth narrows if the specified distortion rate is lowered, and becomes wider if a higher distortion rate is specified.

Expression example:

- Output bandwidth 5Hz~60kHz (Output 50W, 0.05%, 8Ω)

(6) Line Frequency Response

This is the amplifier's frequency response for inputs from line components such as CD players, tuners and cassette decks, and indicates the output level deviation at various frequencies based on a reference output level at 1 kHz. It is generally considered sufficient if the output level deviation is within -3dB at frequencies 1/10 and 10

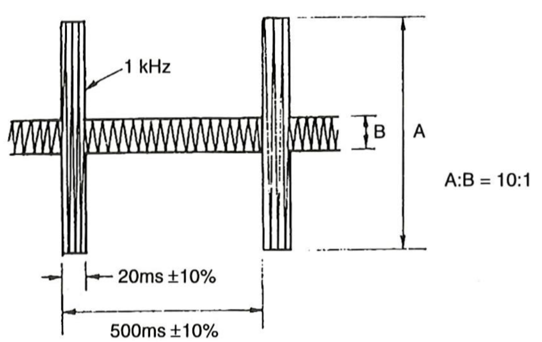


Figure 25 Dynamic Power Measurement Signal

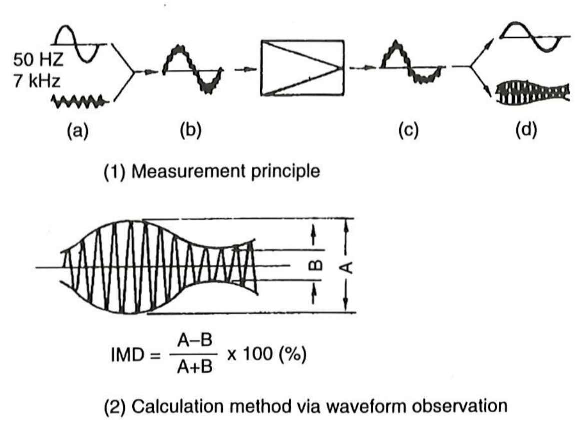


Figure 26 Measurement of Inter-modulation Distortion

* Dynamic headroom: Headroom can be variously defined as, for example, the distance between the top of a door frame and the door sill, or the vertical distance between the ceiling and floor of an attic, and in general expresses a height which allows passage and does not induce a restricted or cramped feeling. With regard to an audio amplifier, if the amplifier's rated output is thought of as the top of a person's head, dynamic headroom expresses how much space there is above the person's head, or in other words, how much surplus power is available above the rated output.

times that of the 20Hz~20kHz audible frequency range.

Expression example:
- 2Hz~200kHz $^{+0}_{-3}$ dB

(7) Equalizer Deviation
Equalizer deviation is a method used to express the frequency response of the equalizer amplifier. The value expresses the equalizer amplifier's level deviation within a specified frequency range corresponding to the RIAA playback equivalence characteristics. Deviation is measured with 1 kHz set at 0 dB.

Expression example:
- 20Hz~200kHz ±0.2 dB

(8) Inter-modulation Distortion
Inter-modulation distortion is the term for the extraneous frequencies and harmonics produced when two or more signals with different frequencies are introduced into an amplifier having nonlinear distortion and inter-modulated. Since an actual music signal is comprised of many different frequency components, the presence of inter-modulation distortion can cause significant loss of sound quality.
The method used for measuring inter-modulation distortion, shown in Figure 26, is as follows: Low and high frequency signals are mixed at a ratio of 4:1 and input into the amplifier. The portion of the output signal produced by modulation of the high-frequency signal with the low frequency signal is measured as the distortion.

(9) Damping Factor
The damping factor is defined as a specified load impedance (8Ω) divided by the amplifier's output impedance. Since nearly all modern day speakers are designed for voltage driven operation, a larger damping factor is usually better. Considering speaker wire and connector contact resistance, however, a damping factor of about 50 or more is good, but a damping factor much higher than this does not provide much extra benefit.

(10) Signal-to-Noise Ratio
An amplifier's signal-to-noise ratio is as important a specification as distortion and output power. There are two methods used for expressing an amplifier's signal-to-noise ratio: the method specified by the EIAJ (EIA uses virtually the same method) and the method specified by the old IHF standards (no longer in existence). While both are used, the old IHF method is still used by a significantly larger number of makers than the EIAJ method.
The method for measuring signal-to-noise ratio as specified by the EIAJ is to input a 0.5V signal as a LINE input, a 5mV signal as a PHONO MM input, and a 0.5mV signal as a PHONO MC input, adjust the volume so that the output for each signal is equivalent to 1W into an 8Ω load (approx. 2.83V), then measure the ratio of the signal voltage to the noise voltage to determine the signal-to-noise ratio. Also included in the measurement setup are input terminal resistances of 1kΩ for the LINE and PHONO MM inputs and 10Ω for the PHONO MC input, as well as an A network for audible compensation.
In contrast to the EIAJ method, the old IHF method measures the ratio of signal voltage to noise voltage when the output power is equivalent to the rated output with the volume set at the MAX position. Measurement is carried out using A network audible compensation with all input terminal resistances set to 0Ω.
Of the signal-to-noise ratios obtained using these two methods, the value obtained using the old IHF method is generally larger than the EIAJ value, which is one reason why EIAJ values still do not appear very often in product specification listings.
When the signal-to-noise ratio obtained with the IHF method is used, true performance may be difficult to judge from the S/N value only. This is especially likely for the PHONO MC input, because the MC input sensitivity setting range is quite large (0.05mV~0.3mV), which means that the output noise level varies widely depending on the input sensitivity, causing the signal-to-ratio to appear worse when the input sensitivity is higher. This situation is illustrated in Figure 27. The signal-to-ratio in the top figure looks better, but the actual signal-to-noise ratio is the same for both cases.

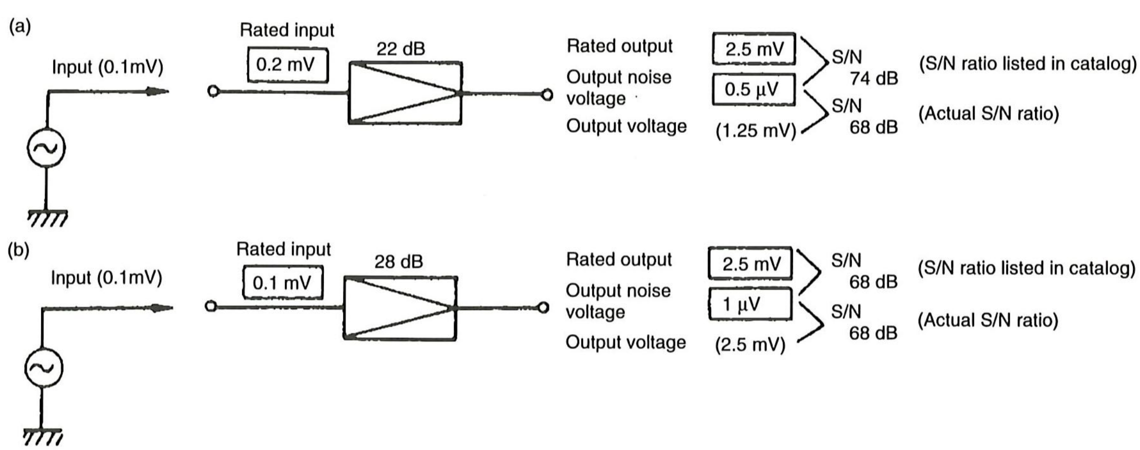
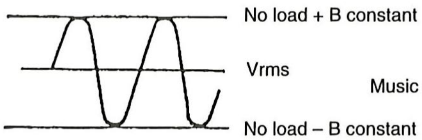


Figure 27 Difference Between Listed S/N Values and Actual S/N Values

(11) Music Power (IHF)
Music power is defined as the maximum output power that can be obtained with no load while holding the secondary voltage at a constant level, assuming ideal transformer regulation.
Unless otherwise specified, output is recorded under the following conditions:
Load: 8Ω, Distortion: 1%, Frequency: 1kHz
Catalog expression method: aW + aW (1%) or 2 × aW (1%)



$$\text{Music power (MP)} = \frac{\text{Vrms}^2}{\text{RL}(8\Omega)}$$

Also, peak music power is defined as twice the music power (MP).
Peak music power (PMP) = $(\sqrt{2}\text{Vrms})^2/\text{RL} = 2 \times (\text{MP})$

(12) Other Specification Items
The above items (1)~(11) are specifications commonly listed in each company's brochures and catalogs. In addition to these, there are many other ways to measure performance, such as channel separation and through rate.
Also, when making catalogs for an overseas market, we have an obligation to list certain values obtained according to the methods used in that particular country. Typical examples are FTC power in the U.S. and DIN power in Germany (Europe). In virtually all cases, the value obtained is lower than the "rated output" value obtained as described in (1). In the case of FTC power, the lower value is due to the fact that after the amplifier is run for an hour at one-third the rated output, it must operate at full rated output for five minutes while not exceeding the rated distortion level over the range of frequencies specified by the manufacturer. In the case of DIN power, the power rating is the maximum power level that can be continuously output for 10 minutes at the specified distortion rate. In both cases, since power is output for a long period of time, the power source transformer heats up causing the winding resistance to increase, thus decreasing the voltage supplied to the amplifier and reducing the maximum amount of power which can be obtained.

4. Key Technologies

The ideal amplifier is a device which will amplify a music source signal to a power level sufficient to drive speakers without altering the music signal in any way, to perfectly reproduce the musicality and atmosphere of the source within the listening room.
The manner and degree to which an amplifier deforms the music signal is significantly influenced by three elements: circuitry, construction and materials (parts).
Accordingly, fundamental amplifier technology is based on studying these three elements from every conceivable angle and eliminating any factor which degrades the music signal.
The new technologies described in this section were developed for the express purpose of eliminating signal degrading factors.

(1) Signal Path
The longer the wiring is inside an amplifier, the more likely it becomes that the signal will pick up external noise or microcomputer clock noise, or that cross-interference between signal lines will occur.
Minimizing the length of the signal path is therefore a basic objective for preventing cross-interference and contamination from undesired signals such as high-frequency noise.

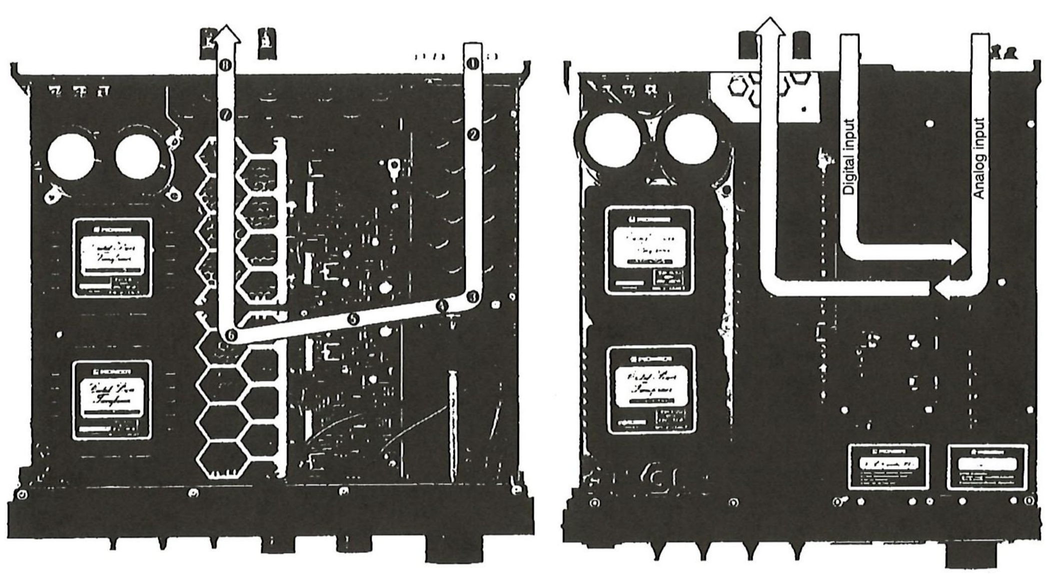


Figure 28 Signal Path Examples

Design of modern day amplifiers is based on this type of thinking, resulting in signal path layouts like those shown in Figure 28. Signal paths in each block, as well, are made as short as possible, resulting in a construction where not only the input and recording selector switches but the main volume control as well extend into the input block, eliminating the need to extend the signal lines to the front panel (Fig. 29).

(2) Local Power Off

The "local power off" concept involves cutting the power to or stopping operation of unnecessary circuits so that only the circuits required for playback of the currently selected source operate.

An example of why this is necessary is the effect of the phono equalizer on other circuit blocks. When an MC cartridge is used, the phono equalizer must operate to amplify a minute 0.1~0.2mV signal all the way up to 150mV to match the line level, requiring an amplification factor of approximately 60dB. Thus, if noise enters the input side of the phono equalizer, the noise will be amplified and become very large, adversely affecting other signal lines through the power source and ground paths. Random noise generated in the phono equalizer circuit when no signal is present can have the same bad influence. It is therefore desirable to cut the power to the phono equalizer circuit when it is not being used.

This type of thinking is of particular benefit in modern day amplifiers to improve the playback sound quality of high signal-to-noise ratio digital sources by designing in various local power off functions for uses such as stopping the digital circuit's system clock or turning off the video power supply.

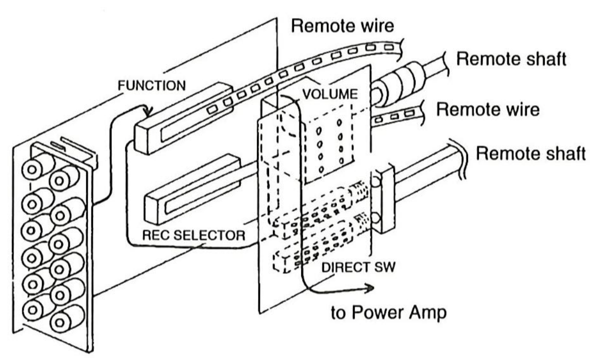


Figure 29 Input Block

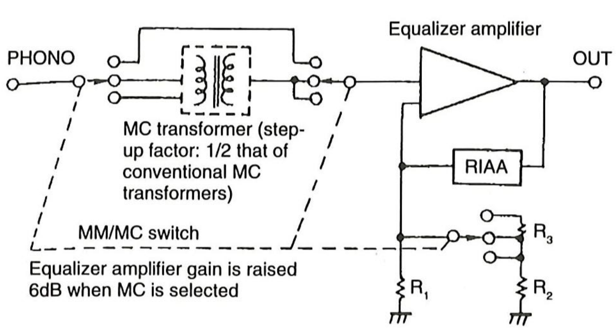


Figure 30 Hybrid MC Transformer Type Phono Equalizer

(3) Hybrid MC Equalizer Circuit

Figure 30 shows a block diagram of a hybrid MC transformer equalizer circuit. The gain of the MC transformer is held to a relatively low 14dB. The gain of the equalizer amplifier used in conjunction with the MC transformer is 36dB in MM mode, and is raised to 42dB for MC use.

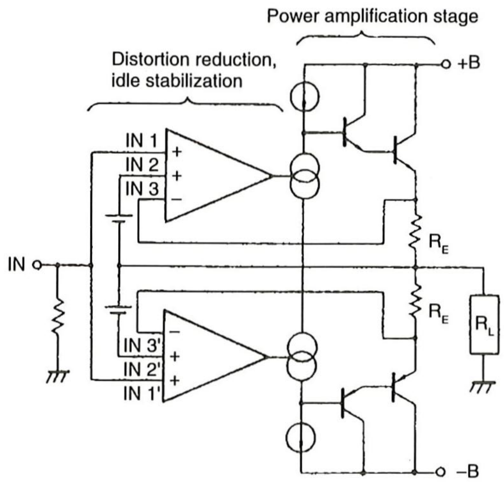
Since conventional MC transformers have a step-up factor greater than 10, a large number of windings are required on the transformer's secondary side. This results in a large amount of distributed capacitance, making it difficult to obtain a flat frequency response in the high range. With the hybrid MC transformer circuit, however, the step-up factor can be set less than 10, thus cutting the number of secondary windings in half. This significantly increases the degree of freedom available in designing the MC transformer, enabling performance improvements through various design measures such as the following: ① use of a part-winding design to reduce distributed capacitance, ② increase in the number of primary windings to improve low-end frequency response, and ③ increase in the turn diameter to reduce DC resistance.

When compared to an equalizer amplifier with a built-in MC head amplifier, it is easier to counteract the signal-to-noise ratio degradation caused by the 6dB increase in equalizer amplifier gain than to suppress the noise generated by the head amplifier.

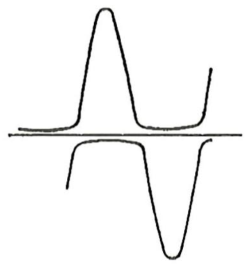
Due to the various benefits just described, the hybrid MC equalizer circuit enables both MC transformer performance improvements and a high signal-to-noise ratio.

(4) Non-Switching Circuit

There are two main types of power amplifier circuits, providing either class A operation or class B operation. Class A operation circuits feature high quality sound reproduction with regard to distortion and other factors, but to achieve this they sacrifice power efficiency and are therefore not suited for use in an amplifier's output stage. To overcome this problem, the "non-switching circuit" (NSC) was developed. A non-switching circuit takes advantage of the high power efficiency of class B operation while reducing the distortion caused by transistor switching, achieving performance on a par with class A operation circuits. This section describes the operation of the second and third generations of this circuit—the non-switching circuit type II and non-switching circuit type III (hereafter referred to as simply type II or type III). Basic operation of the non-switching circuit is shown in Figure 31, and distortion waveforms for various amplifier circuits are shown in Figure 32.

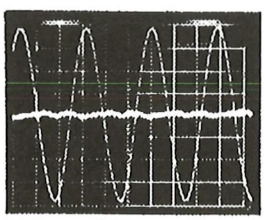


(a) NSC block diagram

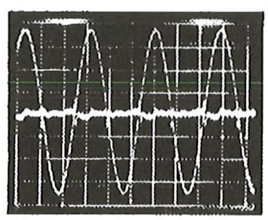


(b) Emitter current waveform

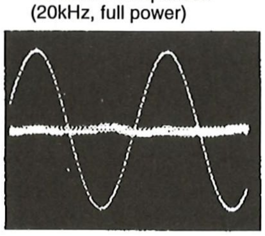
Figure 31 NSC Operation



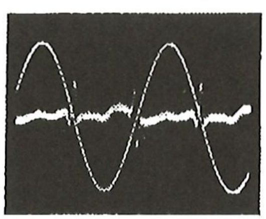
(a) Non-switching amplifier Distortion comparison (20kHz, full power)



(b) General class B amplifier (Pioneer)



(c) Non-switching amplifier [enlargement of (a) waveform]



(d) General class B amplifier [enlargement of (b) waveform/Pioneer]

Figure 32 Distortion Waveforms

The features of these non-switching circuits are as follows:

① With previous circuits, immediately after the power is turned on or when a large signal is input, the output stage idle current does not die out but fluctuates due to changes in the external temperature or the amplifier's internal air flow. With type II and type III circuits, however, the current is stabilized as soon as the power is turned on and will not fluctuate even in the presence of the above-mentioned disturbances, thus constantly maintaining a fixed value.
As a result, generation of temperature-related transient distortion (circle distortion) caused by the above disturbances can be minimized.

② When a type II circuit is voltage driven, output stage distortion can be reduced to less than 1/30 (same for type III).

③ The need for temperature-compensating varistors or transistors is eliminated.

④ Idle current adjustment is unnecessary, and there is little variation in the idle current values between units.

Following is a general description of how a type III NSC operates, based on the diagram in Figure 33. As operation of the positive and negative sides are the same, an explanation is provided only for the positive side. To simplify the explanation, the various base currents inside the IC are ignored. Also, the current mirror made up of Q_1 and Q_2 is assumed to have a current ratio of 2:2, and Q_3~Q_5 are assumed to have identical characteristics.

① When there is no input signal

The base voltage of both Q_3 and Q_4 is 0V. Thus, $I_3 : I_4 : I_5 = 1 : 1 : 2$, and Q_5's V_{BE} becomes slightly higher than Q_4's V_{BE}. As a result, the potential at point A becomes slightly higher than the base of Q_4 (i.e., the output). The value of this voltage divided by R_E is the idle current.

② When a large positive signal is input

Input of a large positive signal causes a positive output current to flow, so the voltage drop across R_E becomes relatively large and Q_4 (Q_6 on the negative side) turns off. As a result, $I_3 : I_4 : I_5 = 2 : 0 : 2$. The state of operation at this time is symbolically illustrated in Figure 34 (a), with the potential at point A equal to the input voltage. As can be seen from the figure, 100% negative feedback is applied, and the signal is transmitted with low distortion.

③ When a large negative signal is input

Input of a large negative signal causes the base of Q_3 (i.e., the base of Q_6) to lower to the same potential as the base of Q_8, so Q_3 (Q_7 on the negative side) turns off, and $I_3 : I_4 : I_5 = 0 : 2 : 2$. The state of operation at this time is shown in Figure 34 (b). Since $I_4 = I_5$, the base voltages of Q_4 and Q_5 are equal. Thus, the potential at point A is equal to the output, so the current flowing through R_E becomes zero. However, if this current really becomes zero, the output will effectively be cut off and non-switching operation will not be achieved. In actuality, therefore, the circuit settings are made so that I_5 (= I_2) is just slightly larger than I_4 (= I_1). By doing this, V_{BE5} becomes slightly larger than V_{BE4}, and the difference in their potential is dropped across R_E. Thus, since the current formed in R_E by this potential drop flows in the output stage, it is possible to avoid cutting off the output.

④ When a small signal is input

Q_3~Q_5 all turn on and the circuit operates between the two operation states just described.

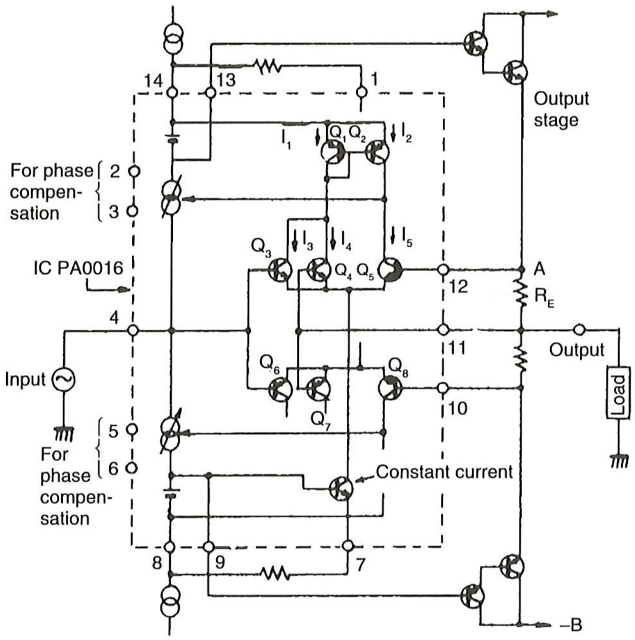
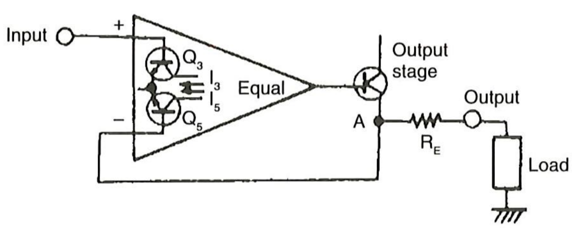
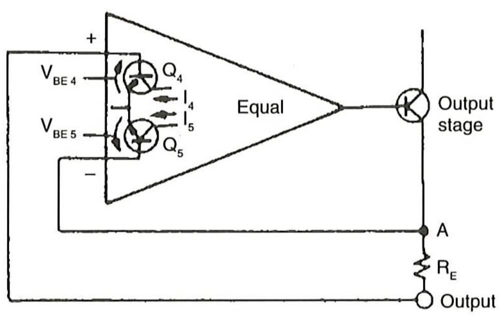


Figure 33 Non-Switching Circuit Type III Principle of Operation



(a) When a large positive signal is input



(b) When a large negative signal is input

Figure 34 Symbolic Diagrams for Explanatory Purposes

(5) Super Linear Circuit (SLC)

Modern day audio amplifiers use mainly transistors as amplification devices. These transistors have nonlinear input and output characteristics (Fig. 35). Because of this, it is necessary to make circuit modifications and apply feedback in order to reduce the distortion this nonlinearity induces. The super linear circuit takes a semiconductor having nonlinearity and connects it with a reverse connection to another semiconductor having nonlinearity of the same mode. This design cancels out the generated distortion, effectively producing a linear circuit.

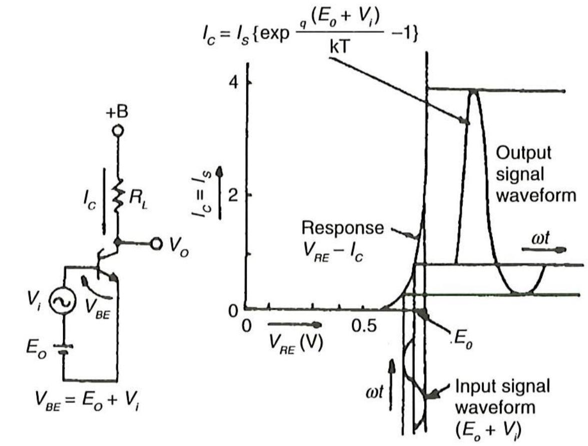


Figure 35 Transistor Input/Output Characteristics

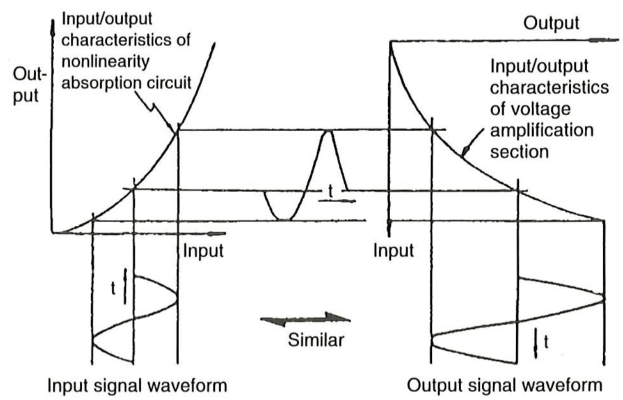


Figure 36 Super Linear Circuit Principle of Operation

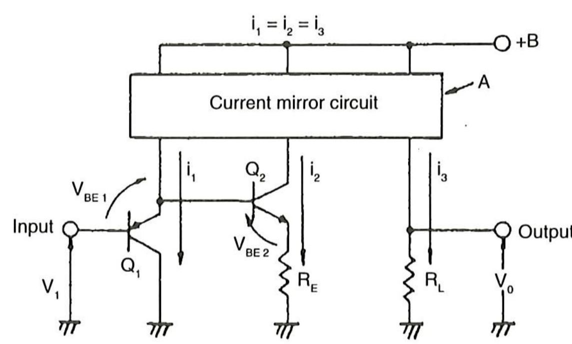


Figure 37 Basic Super Linear Circuit

The operation principle of the super linear circuit is shown in Figure 36, and a basic super linear circuit is shown in Figure 37. The currents flowing through Q_1 and Q_2 are equal due to the current mirror circuit construction, with the result of making V_{BE1} and V_{BE2} equal and effectively canceling V_{BE} induced distortion.

The following relationships are obtained from Figure 37:

$$i_1 = i_2 = i_3$$
$$v_i = R_E i_2$$
$$v_0 = R_L i_3$$

Thus,

$$\frac{v_0}{v_i} = \frac{R_L}{R_E}$$

The features of this circuit are as follows:
① A low distortion rate can be obtained without using negative feedback.
② Circuit stability is superior to negative feedback circuits (in terms of oscillation, etc.).
③ Since circuit stability is good, phase compensation for preventing oscillation is unnecessary, enabling flat frequency response all the way into the high range.

(6) High-Efficiency Class A Amplifier
Power amplifier circuits are classified as either class A operation or class B operation. Class A amplifiers feature high quality sound reproduction with regard to distortion and other factors, but due their poor power efficiency as shown in Figure 38, are not well suited for use in an amplifier's output stage. Thus, class B amplifiers are generally used due to their high efficiency and ability to produce large output power. The non-switching circuit was developed from a class B circuit to reduce distortion while maintaining high efficiency. In addition to this, another type of circuit has been developed to increase the efficiency of class A operation. This circuit is called a high-efficiency class A circuit.
Figure 39 shows a block diagram of this circuit. There are very few modifications added to the original class A amplifier configuration. The main differences appear in the power supply block. This block consists of power source B_1 for supplying the class A operation idle current, and power sources B_2 and B_3 which vary the voltage level according to the input signal level. Since the idle current must only be equal in amount to the output signal, the voltage of B_1 (the voltage between points b and c) is small, so power consumption and heat radiation are also small. Voltages corresponding to the input signal are supplied to points b and c, which are driven in sync with the output signal (point a). The voltage waveforms of points a, b and c are shown in Figure 40. In this way, an amplifier can utilize class A operation and still maintain good efficiency.

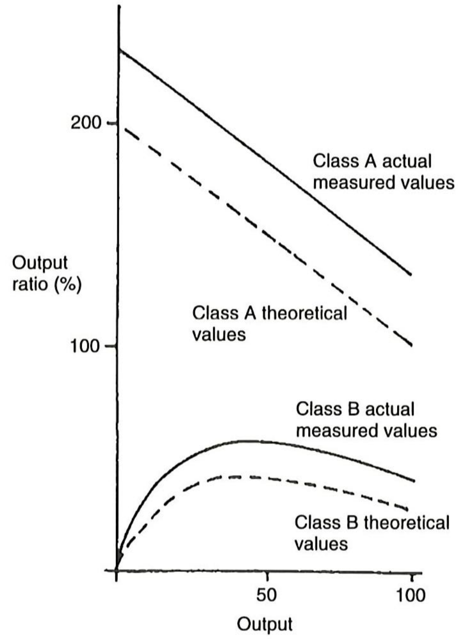


Figure 38 Class A & Class B Loss Curves

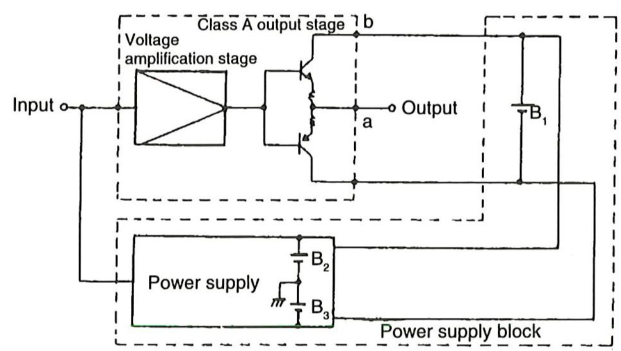


Figure 39 High-Efficiency Class A Circuit Block Diagram

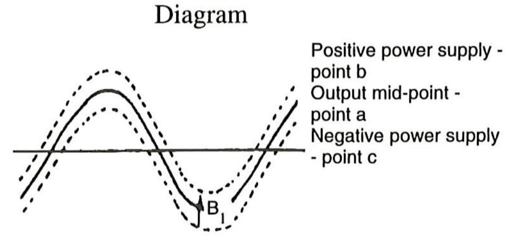


Figure 40 Relationship Between Output Voltage and Class A Circuit Power Source Voltage

(7) Power Supply Block

i. Power Supply Circuit
Two methods of constructing a ±B power supply for a transistor amplifier are shown in Figure 41. As shown in Figure 42, the –B winding of a general power transformer starts near the iron core and thus has a smaller winding diameter than the +B winding, which means that even if both windings have the same number of turns, the lengths of the windings will be different,

resulting in different impedances between the positive and negative power sources.
However, if two independent transformers are used for the + and – sides as shown in Figure 41 (b), it is possible to make the number of turns and winding lengths exactly the same for both sides, thus equalizing the power supply transformer impedances, balancing the ripple current and minimizing unnecessary ground return current.

ii. Power Supply Polarity

It is commonly known that sound quality can be changed significantly by reversing the polarity of the power plug. The reason for this is closely related to the construction of the power cord. Of the two wires that supply electricity to each household AC outlet, one wire is always grounded at the transformer on the power line pole outside. Power cords pick up all kinds of harmful noise such as automobile ignition noise, motor noise and digital noise from computers or devices incorporating microprocessors. There is a possibility that this noise could pass through an amplifier's power transformer and affect the signal circuit.
The power transformer's primary winding begins next to the transformer's iron core, and the core is connected to the component chassis.
In Pioneer products, harmful external noise is prevented from affecting the signal lines inside the amplifier by connecting the ground side of the power cord to the start of the power transformer's primary winding and preventing unnecessary potential from being generated in the chassis (Fig. 43).

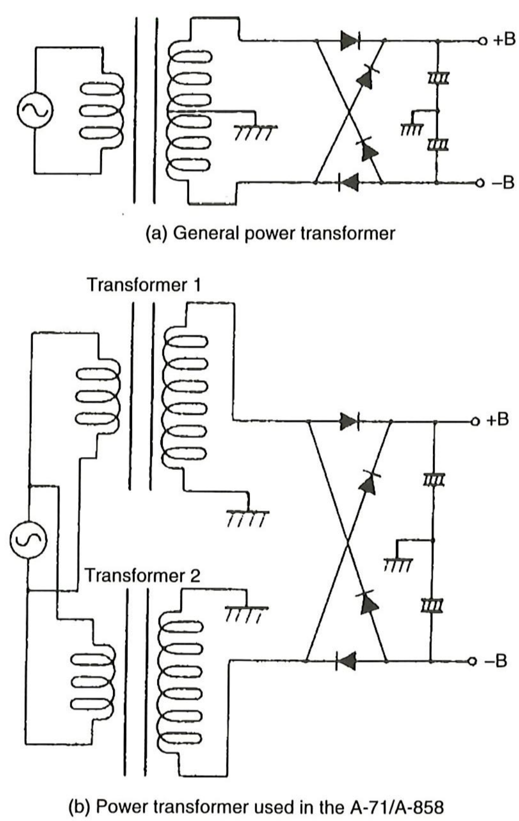


Figure 41 Power Supply Circuit

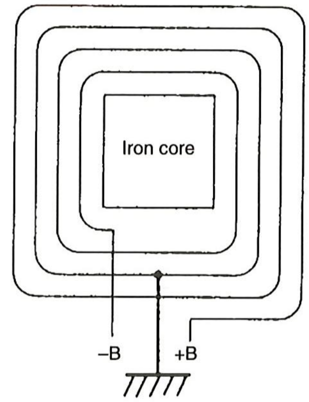


Figure 42 Transformer Winding Example

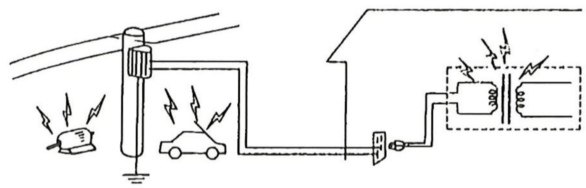


Figure 43 Noise Sources and Propagation Paths

(8) Circuit Components

The circuit components selected for use in an amplifier have a significant effect on sound quality has been an important issue since the late 1970s. It is commonly held that better sound quality can be obtained using nonmagnetic material rather than magnetic material for the lead wires in primary amplifier components such as resistors, capacitors and semiconductors such as transistors.
For capacitors, electrolytic type capacitors are generally used when relatively large capacity is required. The manufacturing process used to produce the aluminum foil wound inside these capacitors, however, significantly affects the resistance of the equivalence circuit shown in Figure 44. For audio use, special capacitors have been developed by reducing the capacitor's "r" value and employing winding modifications and a special pole construction to reduce the inductance "L".
In recent years, much attention has been paid to prevention of induced noise by measures such as electrically shielding the aluminum casing of capacitors to prevent internal capacitor noise from escaping to the outside and preventing external noise from being picked up by the capacitor. (Fig. 45)
Many advances are also being made in the use of nonmagnetic material for wiring, connectors and relays.
Other measures used for maximizing sound quality include the use of large gauge wire or pure copper wire in locations where large current must flow, such as in the output stage, power supply circuit and ground lines, and the use of gold-plated terminals for connectors such as PHONO jacks which transmit low-level signals.
Steel is generally used for the amplifier chassis, but since it is a magnetic material, generation of small amounts of magnetic distortion in the signal is possible if the signal

line runs close to the chassis. Measures used to prevent this include keeping the signal wires away from the chassis, plating the chassis with copper or manufacturing the chassis from a non-magnetic material such as aluminum or plastic.

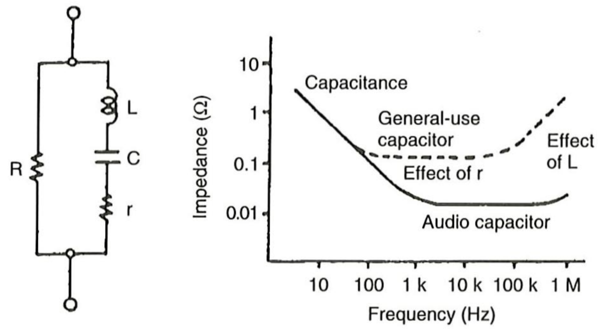


Figure 44 Capacitor Equivalence Circuit and Characteristics

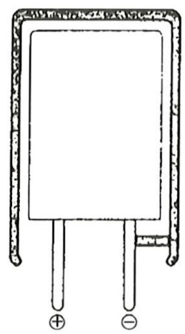


Figure 45 Shielded Capacitor Construction

(9) Mechanism Technology

The main sources of vibration in an amplifier are the power transformer, electrolytic capacitors and power transistors, and common sources of vibration from outside the amplifier include sound pressure from speakers and vibrations from other components with rotating parts. These vibrations are transmitted by the cabinet and chassis and finally cause the wiring and parts on the printed circuit boards to vibrate. These vibrations cause changes in the contact resistance and induced voltage of resistors, capacitors, switches and wiring, generating various types of distortion and thus adversely affecting sound quality. It is therefore necessary to construct each unit to block vibrations from entering the unit or to instantaneously damp vibrations that do enter the unit. Methods for increasing body rigidity and blocking vibrations are explained below.

i. Honeycomb Pattern for Improved Rigidity

Four types of manufacturing processes are used for producing honeycomb patterned parts, as follows:

① Drawing (frame parts)
② Aluminum extrusion process (heat sinks)
③ Press punching (cabinet)
④ Plastic molding (legs)

The common factor among the various parts made with these processes is their honeycomb pattern which disperses received vibrations in many directions to instantaneously damp them and prevent them from affecting other parts. The honeycomb pattern also serves to increase the rigidity of the parts.

By constructing an audio amplifier with honeycomb patterned parts, high rigidity and good vibration damping characteristics can be achieved. (Fig. 46)

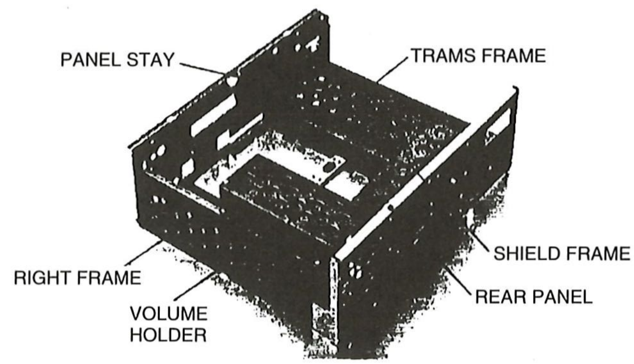


Figure 46 Honeycomb Frame Chassis

ii. Use of Anti-Vibration Materials

Prevention of vibrations over a wide range of frequencies is possible if the actual material used for the various parts has the ability to quickly dissipate vibrations.

One example of this is the "casted power transformer," which is made by placing the biggest source of vibration of an amplifier—the power transformer—into a cast iron casing and filling the casing with silica-reinforced polyester resin. Cast iron is used because of its high rigidity, high internal loss and good vibration damping characteristics.

A special paper is used to provide a floating mount for the input unit chassis. This paper is made of long fibers elaborately woven to effectively absorb harmful vibrations in the 2kHz~4kHz range. Further anti-vibration measures can include use of brass for the volume control remote shaft and use of microcell polymer sheets for cushioning the legs.

iii. Other Technologies

Use of copper plated parts is beneficial for two reasons: (1) copper plating absorbs magnetic distortion by converting magnetism into eddy current, and (2) lamination of dissimilar metals increases the material's vibration damping characteristics.

Some audio amplifiers have five legs. This design is used to prevent degradation of sound quality which can be caused by slight twisting of the frame—a possibility even with very rigid frames. An especially effective sound quality improvement measure is to place a leg directly under the power transformer to suppress transformer vibrations and prevent the vibrations from being transmitted to other parts of the amplifier.

(10) Digital Technology

In the present day and age, CDs, laser discs containing digital audio and DATs have become common audio sources, and various other types of digital sources are expected to appear in the future. This section describes the digital technology used in audio amplifiers today.

i. Digital Audio Interface

Digital source components such as CD players incorporate a D/A converter (hereafter referred to as a "DAC") for converting digital signals to analog signals for reproduction. In addition to this DAC, digital recording devices such as DAT decks and CD-R players also incorporate an A/D converter (ADC). When these digital sources are combined with audio amplifiers and speakers in conventional audio systems where analog audio signals are transmitted between components via pin cords, source signals may pass through several DAC and ADC units before finally being reproduced, creating a very uneconomical system.
Also, with analog signal transmission there is the possibility of noise such as mechanical drive-related noise, servo circuit noise and RF noise from the source component polluting the signal. The connection cables themselves and input/output connectors may also have an adverse affect on sound quality. Although many digital components are designed with various measures for minimizing these analog transmission-related problems, the easiest method is to avoid the problems altogether by transmitting the signals digitally.
By centralizing DAC and ADC operations in the amplifier and transmitting signals between the source components digitally, the system becomes much more economical and streamlined, and signal degradation is kept to an absolute minimum. A block diagram of a digital audio amplifier is shown in Figure 47.
With this objective in mind, digital audio interface (DAIF) standards (EIAJ: CP-340, CPX-1201, IEC: IEC Publ. 958, etc.) have been defined to ensure good connectability between digital components made by different manufacturers.
DAIF signals are transmitted between components using a self-synchronous serial transmission protocol at one of three standard sampling frequencies: 32kHz (DAT decks), 44.1kHz (CD players, laserdisc players, DAT decks, PCM encoder/decoders) or 48kHz (DAT decks).
Bi-phase modulation is used for the modulation method because it minimizes the transmitted signal's DC component, facilitates clock reproduction from the transmitted data, and places no restrictions on connection polarity.
Audio data transmission is limited to two channels (use of four channels is reserved). One frame consists of a sequence of subframes, with the frame length equal to the sampling period. Two subframes make up one frame and 192 sequential frames make up one block. The start of each block is demarcated by a special subframe preamble. As shown in Figure 48, each subframe consists of 32 time slots. These 32 time slots comprise 4 time slots used for extracting the system clock and obtaining synchronization at the demodulation side [preamble: three patterns (B, M, W)], 20 time slots containing audio data [audio sample word], 4 time slots for extended or auxiliary audio data [audio auxiliary] and 1 time slot each for V, U, C and P data (see Fig. 48).
The V slot is for the validity flag, which is used to indicate whether the value of the audio sample word is correct or not.
The U slot is for user data. User data is data other than the V, C, P or audio sample word, such as data used for transfer of CD subcodes.
The C slot is a very important time slot called the channel status, and is provided for transferring the data of each demodulatable audio channel in a specified format for any interface user. Transferable data includes pre-emphasis data, sampling frequency, time code, source number, category code, channel number, clock precision, copyright protection data and digital copying

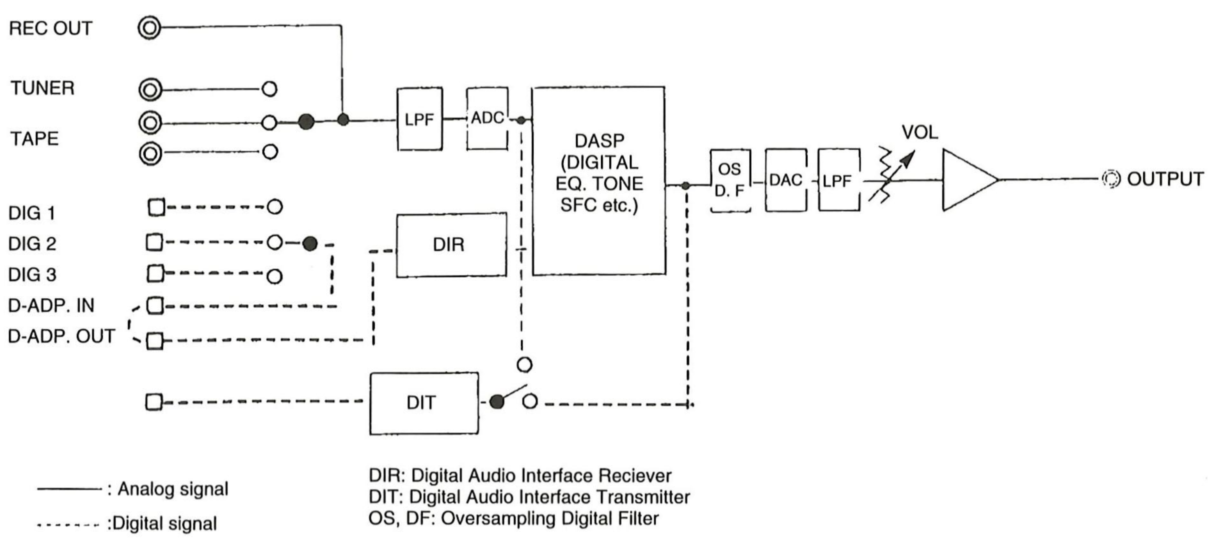


Figure 47 Digital Audio Amplifier Construction

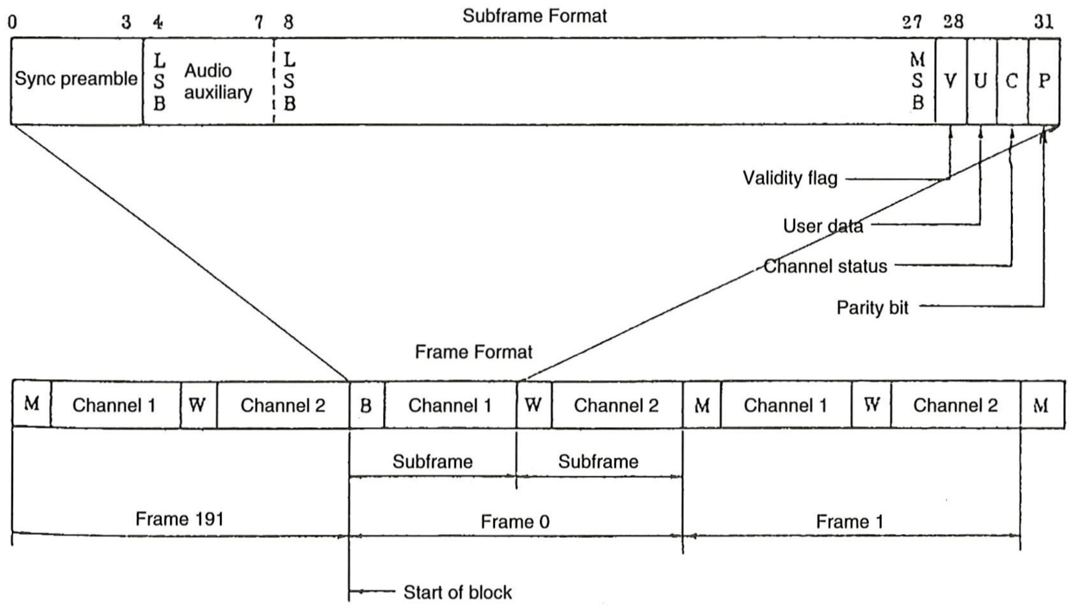


Figure 48 Digital Audio Interface Format (EIAJ CP-340)

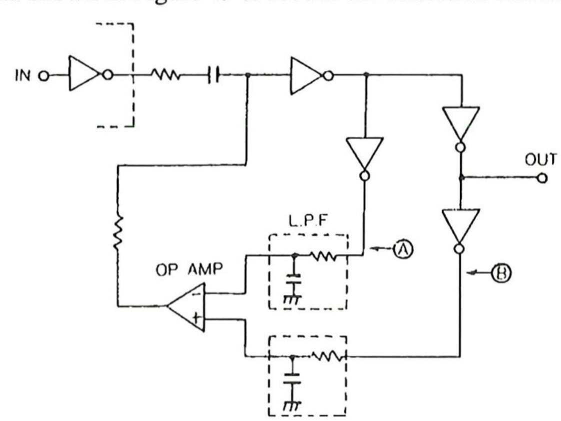
prohibition/permission data (SCMS: Serial Copy Management System).
The P slot is called the parity bit, and is provided to enable detection of odd parity errors caused by interface malfunction.
The DAIF thus has many merits for transferring various types of "sub" information.

ii. Optical Data Transmission

Two line types are available for electrical data transmission: unbalanced lines for consumer-use equipment and balanced lines for professional use. Unbalanced lines use coaxial cable with an impedance of 75Ω pin and equipped with a pin connector (EIAJ, RC-6703), with a transmission signal level of 0.5Vp-p.
In addition to electrical data transmission, digital data can be transmitted optically by modulating optical signals according to the DAIF signal format and transmitting the signals using optical fiber cable. This type of transmission has the following merit:

① Since signals are transmitted between digital components using optical fiber, electrical separation of source components is achieved, eliminating the effects of source component noise.
② There is no unnecessary cable radiation such as with coaxial cables.
③ Effects from power transformer leakage flux are prevented.

On the reverse side of the coin, however, optical transmission also has a demerit: if the fiber is subjected to bending or vibration, the width of the modulated pulse signal fluctuates, altering the duty factor. This causes the clock demodulated by the PLL at the receiving side to fluctuate, generating distortion in the analog signal produced from the D/A conversion process. This distortion in the optical signal pulse is called pulse width distortion. Some digital components incorporate an optical transmission distortion compensation circuit like that shown in Figure 49 to reduce the distortion effects.



Low-frequency jitter is reduced by feeding back the level difference between points A and B and making it equal to the sum of the "H" level and "L" level.

Figure 49 Optical Transmission Distortion Compensation Circuit

iii. Digital Signal Processor

A digital signal processor (DSP) incorporates a multiplier in its hardware configuration for performing high-speed calculations using methods such as pipeline processing, and can be thought of as a type of microcomputer used for processing digital signals in real time.

(a) General DSP features

Like a microcomputer, a DSP processes signals by performing various calculations according to specially written software, thus making it an extremely flexible

tool for realizing different audio functions.
Also, DSP hardware can be made much more compact than analog circuits for the same purpose, and various functions can be added or altered by simply changing the software without making changes to the hardware. Moreover, "adaptive control" processing which is difficult using analog circuitry can be realized relatively easily, and both reproducibility and processing precision are better than with analog processing.
Other features include high stability with respect to temperature changes and changes over time, and the elimination of the need for an adjustment process during manufacturing.

(b) Pioneer's original audio DSP — DASP (Digital Audio Signal Processor)
DSPs can be broadly classified into general-use processors and special-use processors. Pioneer has developed an original special-use processor for audio equipment called the DASP which is used in many currently available products. The features of the DASP (PD0051, PD0055) are as follows:

① High precision processing
If a short calculation word length is used when processing low frequencies with functions such as filters, oscillation or degradation of the signal-to-noise ratio may occur. With the DASP, however, a multiplier word length of 47 bits (data length: 32 bits X coefficient word length: 24 bits) and a word length of 50 bits for the AU (Arithmetic Unit) and accumulator provide high calculation precision to prevent such problems from occurring.

② High-efficiency pipeline processing
High-efficiency processing is realized with the DASP's high-speed calculation speed of 160ns per instruction, three-stage pipeline processing and parallel processing (1 instruction: 31 bits).

③ Built-in external RAM interface
An interface for adding external dynamic RAM is built-in, the delay data word length can be set to 16 or 24 bits and the number of dynamic RAM units can be set from 1 to 3, providing a large degree of design flexibility to meet the needs of various digital audio systems.

④ Various audio interfaces
The audio interface provides two audio inputs (four channels) and three audio outputs (six channels). This feature, combined with selectable data word length (16/24 bits) and LSB first/MSB first options, make it easy to interface with peripheral LSIs.

⑤ All-RAM memory
Since all microprograms, coefficient data and delay data are downloaded into corresponding RAM areas through the microcomputer interface, data can be freely modified and software can be easily replaced to realize various functions.

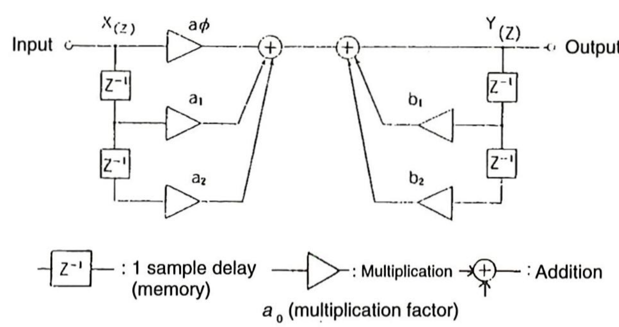


Figure 50 2nd Order IIR Digital Filter

Filter type, characteristics	Coefficient values				
	a_0	$a1$	$a2$	$a1$	$a2$
Bell filter f_0=1 kHz, +10 dB, Q=1.5	1.097718554	−1.89026593	1.761866888	1.890265938	−0.90961516
Bell filter f_0=1 kHz, −10 dB, Q=1.5	0.910980320	−1.72199507	0.828641515	1.721995070	−0.73962183
Pass filter (LPF) f_0=1 kHz, −12 dB/OCT	0.004603998	0.009207997	0.004603998	1.799096409	−0.81751240
Pass filter (HPF) f_0=1 kHz, −12 dB/OCT	0.904152203	−1.80830440	0.904152203	1.799096409	−0.81751240
Shelving filter f_0=500 HzB, +12 dB	3.845589194	−7.42128010	3.580426533	1.862368234	−0.86710386
Shelving filter f_0=500 HzB, −12 dB	0.260038176	−0.48728683	0.225480106	1.929816142	−0.93104758
Shelving filter f_0=500 HzB, +12 dB	1.028590621	−1.72983135	0.750089990	1.798287976	−0.80845991
Shelving filter f_0=500 HzB, −12 dB	0.934874256	−1.74108011	0.810633076	1.734481196	−0.75210625

Table 5 Coefficient Values for Various Filter Types (fs = 44.1kHz)

⑥ High throughput
For example, a second-order IIR (infinity impulse response) filter can be realized with a minimum of five steps.

(c) Functions implemented with an audio DSP
① Audio-use filters
Second-order IIR digital filters like that shown in Figure 50 are commonly used for various audio-use filters including equalizers, loudness control filters, tone control filters, high-pass filters and low-pass filters. IIR filters have a theoretical impulse response of infinity, in contrast to FIR (Finite Impulse Response) filters which have a limited impulse response. The second-order transfer function *H(z)* for this filter is generally expressed as follows:

$$H(z) = \frac{a_0 + a_1 \cdot z^{-1} + a_2 \cdot z^{-2}}{1 - b_1 \cdot z^{-1} - b_2 \cdot z^{-2}}$$

The values of the coefficients $a_0 \sim b_2$ determine the filter's frequency characteristics (refer to Table 5). One method used to determine these coefficients is called bi-linear conversion. The following formula is used to define the relationship between z and s ($s=jw$) of the transfer function *H(s)* for the analog domain:

$$s = \frac{2}{T} \cdot \frac{1 + z^{-1}}{1 - z^{-1}}$$

($T = 1/fs$, (fs: sampling frequency)
② Sound field simulation
When listening to music performed live in a concert hall, etc., the listener hears not only the sound transmitted directly from the sound source, but reflected sounds from many different directions as well. (Refer to section 5 (2).) The DSP can be used to recreate these reflected sounds to simulate various sound fields such as concert halls, stadiums, jazz clubs and cathedrals. This type of simulation function is widely used today in audio equipment such as mini (and mini-mini) component systems, receivers, AV amplifiers, karaoke amplifiers and surround processors.
Initial reflected sound processing is carried out as shown in Figure 51 using the DSP's external RAM to temporarily hold a copy of the sound data and add it to the original sound data after a short delay period. Various combinations of the reflected sound level and delay time can be freely set by changing the coefficient data and offset (delay) data, enabling the creation of different sound fields.
The successive reverberation (echo) sound that follows the initial reflected sound can be realized by, for instance, constructing the reverberator circuit proposed by M. R. Schroeder. This reverberator is formed by connecting several comb filters (Fig. 52 (a)) in parallel and summing their outputs, then adding a cascaded connection of several all pass filters (Fig. 52 (b)). The reverberation time can be changed by varying Tc (offset data) or g (coefficient data). The DSP's external RAM is normally used for this purpose also.
③ Other
In addition to filter and sound field simulation functions, the DSP is also used for Dolby Pro Logic surround processing and key control, stereo wide, simulated stereo, audition (vocal canceling), noise reduction, expander/compander, LPS (Listening Point Selection), noise gate and limiter functions. Development of new functions and algorithms is expected for use in future products.

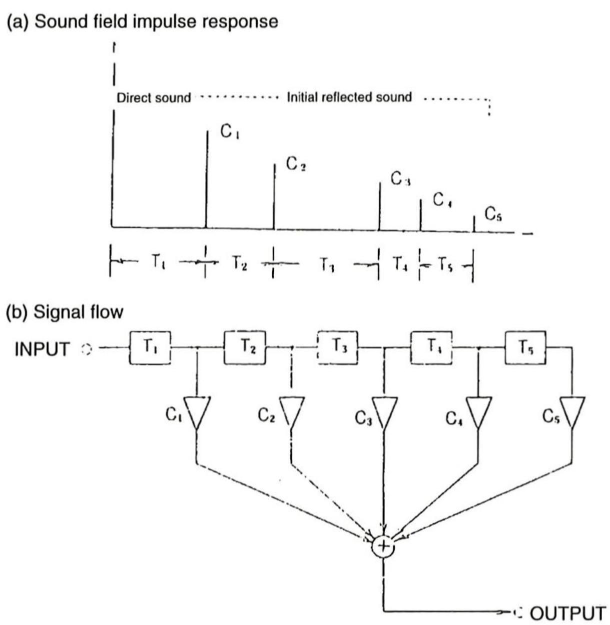


Figure 51 Initial Reflected Sound Processing

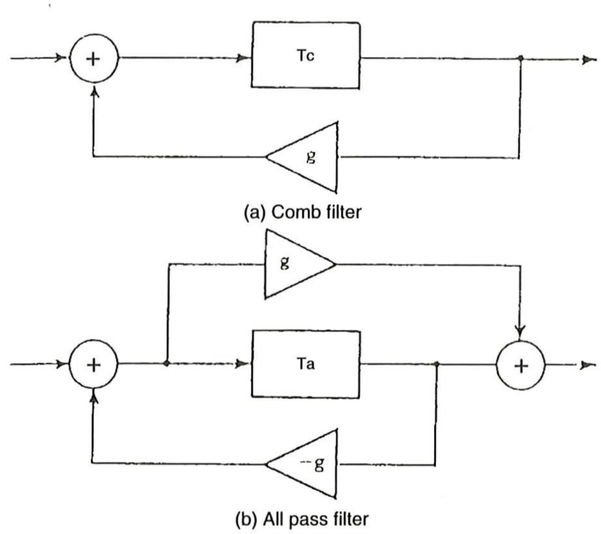


Figure 52 Filters Used For Reverberation Processing

(11) Use of Microcomputers

Microcomputers are used in many audio components today, including amplifiers. In an audio amplifier, a microcomputer is used for controlling various circuits such as switching circuits (e.g., function switches), volume control circuits, protection circuits and display circuits.

Use of a microcomputer provides many benefits, including the following:

① Signal switches can be freely positioned within the circuitry, allowing the length of the signal path to be reduced.

② Operation switches can be freely positioned on the front panel, allowing greater design freedom.

③ Special timing functions that would be very complex if implemented using hardware can be created relatively easily.

④ Remote control functions are possible.

(12) Liquid Crystal Television Displays

As modern day audio amplifiers incorporate an increasing number of functions and AV features, improvement in ease of use is becoming an increasingly important factor. One means for realizing this is the use of liquid crystal display (LCD) technology for providing clear indication of component operation status in a multi-function centralized display panel.

Types of liquid crystal displays include simple matrix displays, TFT active matrix displays and MIM active matrix displays, classified according to differences in the methods used to drive the liquid crystal. In general, however, all of these display types operate in the TN display mode. In a TN display mode LCD, liquid crystal molecules sandwiched between two glass sheets are oriented so that their direction of orientation rotates 90° from one glass sheet to the other (Fig. 53 (a)). Accordingly, light passing through the display is turned 90° along the molecular orientation of the liquid crystal molecules. If an electric field is applied to the liquid crystal layer, however, the dielectric anisotropy of the liquid crystal molecules causes the molecules to re-orientate as shown in Figure 53 (b), causing the liquid crystal layer to lose its ability to change the direction of light passing through the display.

By making use of this property and placing a polarizing filter on each side of the liquid crystal layer, it is possible to control the passing of light through the liquid crystal by turning the electric field on and off. In an LCD television, every pixel is operated in this manner.

When mounted in an audio component, an LCD television can be used not only for watching videos or TV broadcasts, but also for displaying various functions such as the recording selector position, surround mode, volume level, balance, tape deck operating status, program recording information, telephone reception monitor, and appropriate messages regarding component operation, by using a character generator to display characters and/or picture icons each time an operation is carried out. Different colors can also be used for different types of data to make it easier to distinguish data when viewing from a distance.

Using an LCD television in this manner enables various other display possibilities as well, limited only by the imagination. In the future, even more effective use of the freedom of expression made possible with LCD televisions can be expected.

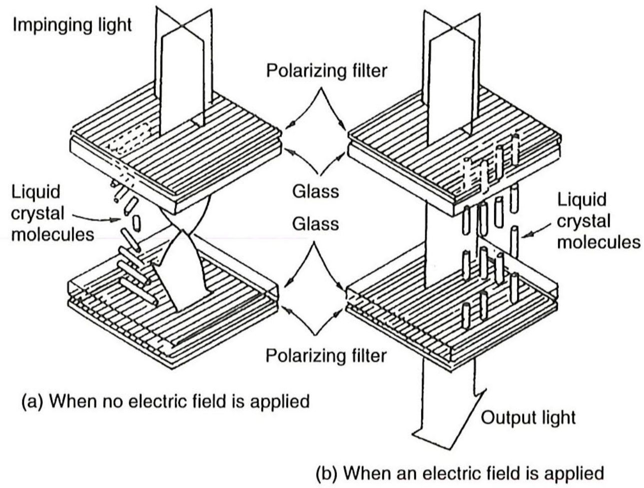


Figure 53 TN Liquid Crystal Operation

5. Peripheral Components

(1) Graphic Equalizer

A graphic equalizer is a tone control device which divides the audio frequency range into several bands and allows the user to independently increase or decrease the level of each band. The characteristics of a typical graphic equalizer are shown in Figure 54.

The main uses and functions of a graphic equalizer are as follows:

① A graphic equalizer can compensate performance variations (generated by mechanical resonance, etc.) of electrical circuits and electrical-mechanical conversion components.

② Fine adjustments can be made to compensate for the acoustic transmission characteristics of the listening room.
For example, a test record, CD or microphone can be used to make fine sound quality adjustments for the overall system including the player, amplifier, speakers and listening room.

③ Fine adjustments can be made to subtle nuances of the music or musical expression to suit the user's individual preferences.

④ A graphic equalizer can be used as a simple filter to cut unnecessary frequencies.

An graphic equalizer circuit incorporating an LC series resonance circuit is shown in Figure 55. In this circuit,

the impedance drops at the frequency f_l determined by L_l and C_l according to the equation $f_l = 1/2\pi\sqrt{L_l C_l}$, and the gain near the f_l can be varied by adjusting the center-grounded potentiometer VR_1. When the slide knob of VR_1 is at the center position, the frequency response is flat; when the knob is moved toward the output side, the gain near f_l attenuates; and when the knob is moved toward the amplifier's reverse input side, the gain near f_l increases.

When using an LC series resonance circuit configuration, a coil can be used for L, but since a core is necessary for generating a large inductance, magnetic saturation may occur, causing adverse effects such as dynamic range reduction or pick-up of electromagnetic induction hum or noise. To avoid this it has become common recently to use "semiconductor inductors" made from a network of resistors, capacitors and semiconductors. The principle behind the semiconductor inductor is shown in Figure 56. Compared to conventional coil inductors, semiconductor inductors are superior in performance and also have the advantage of being easy to implement in IC form.

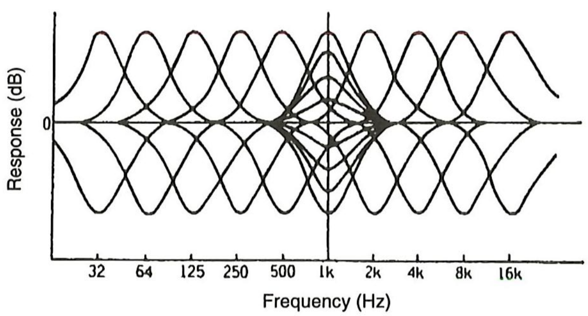
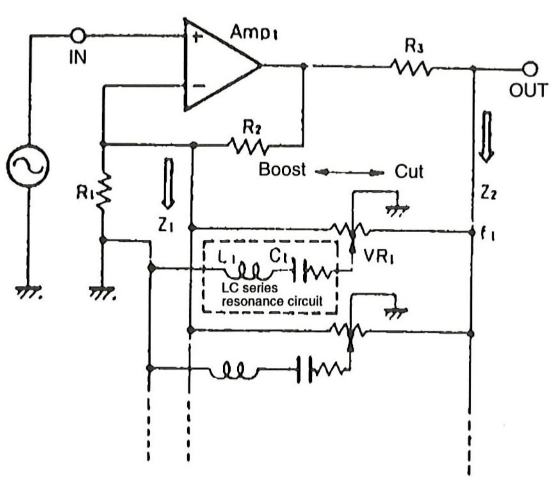
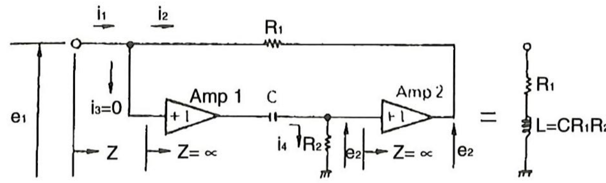


Figure 54 Graphic Equalizer Characteristics



Graphic equalizer circuit using an LC series resonance circuit

Figure 55 Graphic Equalizer



$$Z = \frac{e_1}{i_1} = R_1 + j\omega \underbrace{CR_1R_2}_{L}$$

Figure 56 Semiconductor Inductor Principle

(2) Surround Processor

In concert halls or auditoriums, the sound heard by the listener is comprised of both direct sound from the sound source and indirect sound reflected in a complex manner from the walls and ceiling. A surround processor can simulate and create this same type of sound field to achieve the atmosphere of a concert hall or theater in the listening room.

Four to seven speakers are generally required for the most effective reproduction of signals processed in different ways to achieve various effects.

Since high-quality delay and signal processing functions are required with this type of component, makers are moving away from conventional analog processing methods toward use of digital processing methods.

i. Dolby Surround

Dolby Surround is a surround processing technique for extracting multiple direction data from a two-channel source, and is used in movies to achieve various effects such as moving the sound with the picture or making the viewer feel surrounded by sound. Dolby Surround is possible only if the necessary data is encoded in a specified location on the tape, film or disc.

In the past, Dolby Surround decoders were passive decoders using L-R signals and delays. As movie screens became larger and picture quality became higher, however, active decoders enabling higher separation decoding became the mainstream.

A block diagram of a Dolby Pro Logic decoder is shown in Figure 57.

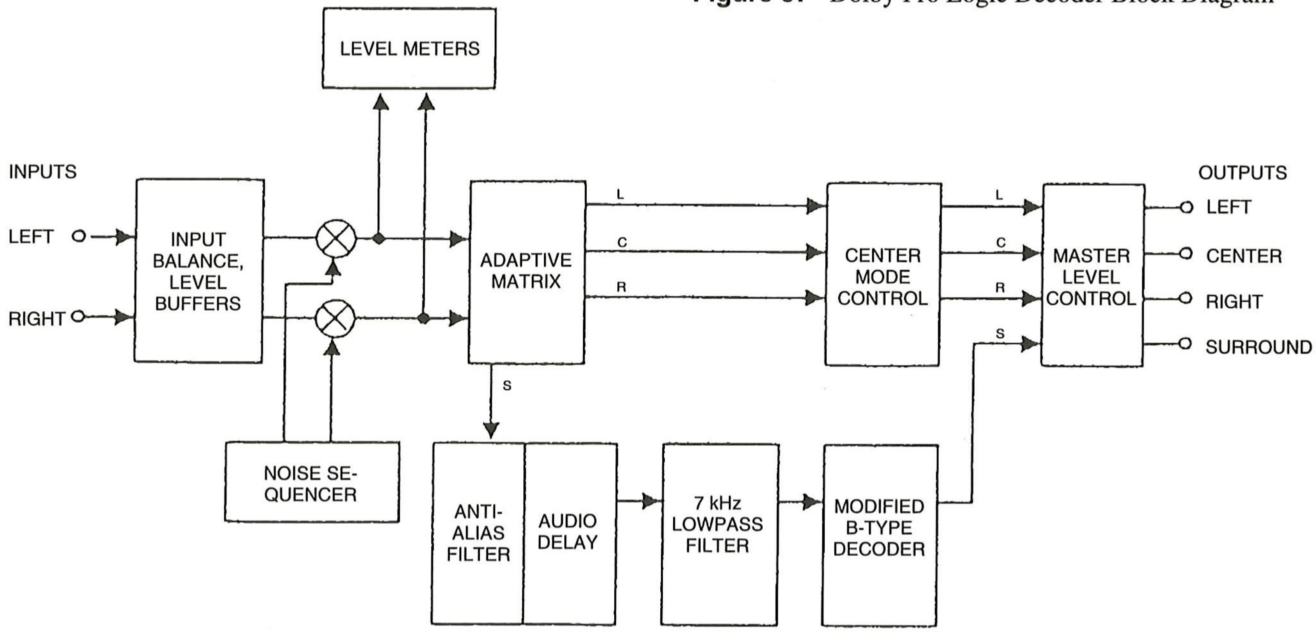
With previous passive decoders, the L-R signal was reproduced using the rear (surround) channel and the L+R signal was reproduced using the center channel, resulting in low separation between front and rear directed sound, and between the center channel and the left and right channels. The Pro Logic decoder, on the other hand, uses an adaptive matrix directionality emphasis circuit (Fig. 58), achieving L-C-R-S separation greater than 25dB.

Operation of the Pro Logic decoder is as follows: The L, R, L+R and L-R components are extracted from the input signal using rectification detection, and the dominant signal is selected. This dominant signal is

subjected to directional emphasis processing by eight VCAs and a signal unification network in the final stage, then is output as L, R, C and S signals. The S signal is output after passing through a delay circuit, 7kHz low-pass filter and modified B-type decoder.

The adaptive matrix, delay circuit, modified B-type decoder and 7kHz LPF can all be digitally processed using a DSP, making it possible to avoid performance degradation caused by use of analog circuit elements.

Figure 57 Dolby Pro Logic Decoder Block Diagram



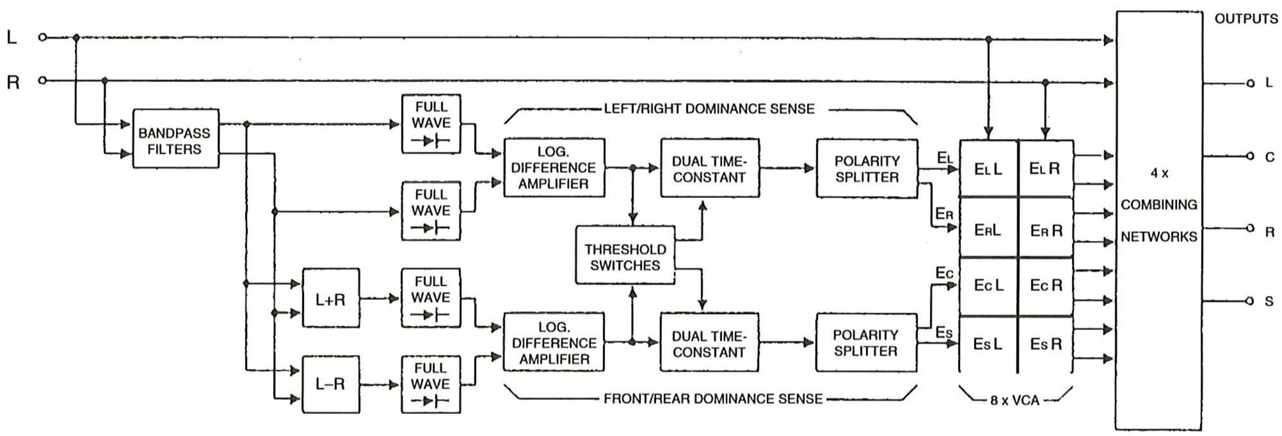


Figure 58 Adaptive Matrix

ii. Sound Field Control

When listening to music in a concert hall or theater, indirect sound reflected from the walls and ceiling of the hall exists in addition to the direct sound from the instruments. Sound which follows the direct sound by a 50~100ms delay is called initial reflected sound, and sound heard after a longer delay is called echo sound, or reverberation. Initial reflected sound provides the live atmosphere and feeling of spaciousness of the hall, and echo sound affects reverberation and sound clarity. (Fig. 59)

These indirect sounds are generated using a DSP and reproduced using several speakers, creating a sound field similar to a concert hall or theater in the listening room. Of particular importance in the generation of sound fields is the number of indirect sound "echoes," the delay time and sound field data such as directionality. Several methods are available for obtaining this data.

Figure 60 shows a sound field analysis generated using computer simulation techniques. This analysis technique is called the sound ray method, and uses computer simulation to obtain reflected sound data from the shape of the hall and the positions of the sound source and reception points (audience).

The signal flow inside the DSP of the SP-91D sound field processor is shown in Figure 61.

The SP-91D uses two high-performance PD0055 DSPs for generating the initial reflected sound and reverbera-tion sound and for providing the function of the two-channel 7-band equalizer.

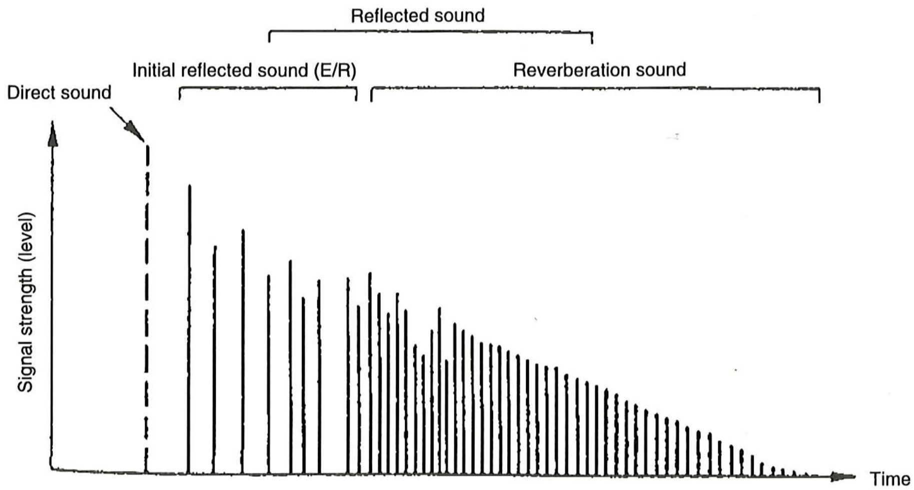
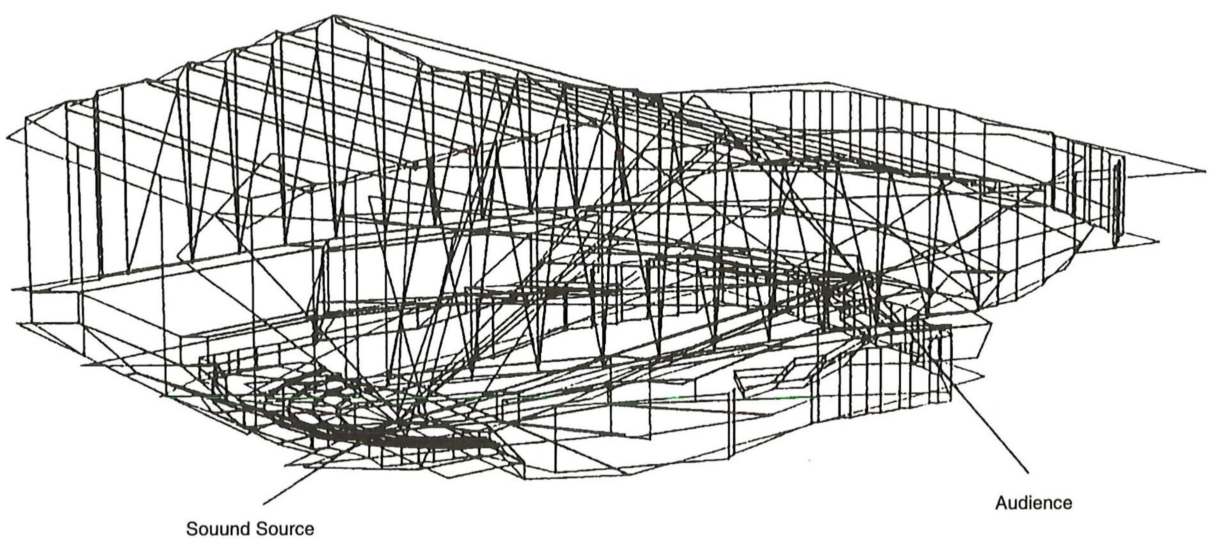


Figure 59 Indirect Sound Groups in a Sound Field



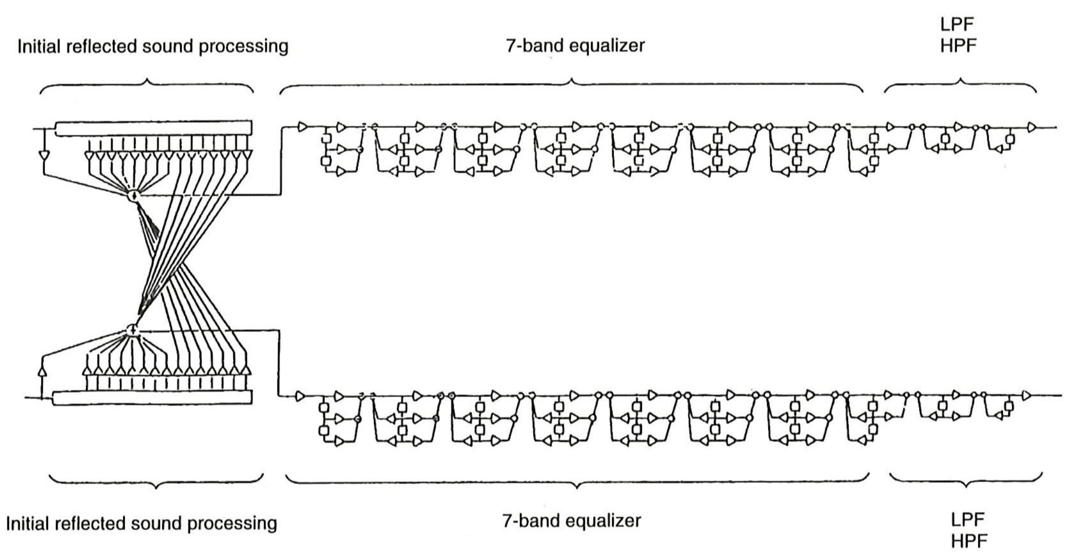


Figure 61 DSP Signal Flow in the SP-91D

iii. Other Surround Functions

In addition to types of surround described above, various other surround modes are also available. Commonly used surround modes are listed in Table 6. (Other types of surround modes may also be provided depending on the product.) Delay time is generally in the vicinity of 15ms~30ms.

The processing used to generate these surround modes, often using L-R with delay, is relatively simple compared to the previously described methods.

Table 6 Surround Modes

Surround mode	Surround L channel	Surround R channel	Remarks
Dolby	*L–R delay	*L–R delay	* With Dolby decoder
Theater	L–R delay	L–R delay	
Stadium	L–R delay	L–R	
Simulated	L+R delay	L+R delay	

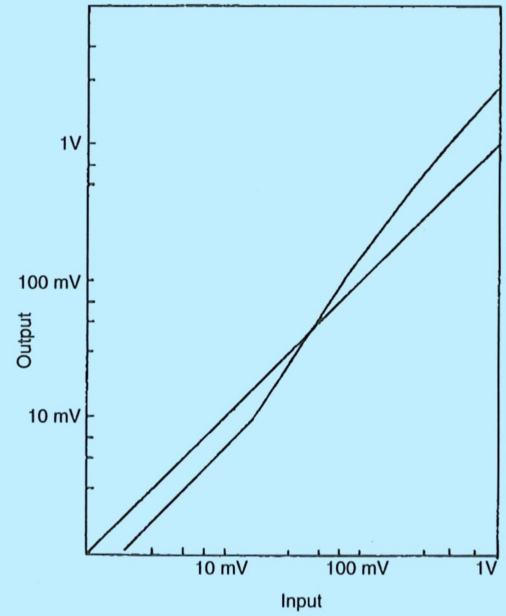


Figure 61 Expander Input/Output Characteristics

(3) Expander

An expander is a component which expands the dynamic range of the program source, effectively producing a more energetic, live-sounding atmosphere and more powerful sound. The input/output characteristics of a typical expander are shown in Figure 61. As can be seen, the characteristics are not a straight line. When the input level is small, the gain is small, and when the input level is large, the gain is large also. Accordingly, when a large signal is input, the signal becomes even larger, thus increasing the power of the sound. At the same time, small input signals are made even smaller, thus providing a noise reduction effect.

Figure 62 shows the operation principle of an expander. The gain control amplifier (GCA) is a negative feedback amplifier which varies the gain according to the control current, and amplifies input e_I to obtain the output e_0. The detector unit detects the input signal and converts it into a DC voltage for generating the DC control current I_a, which is supplied to the GCA.

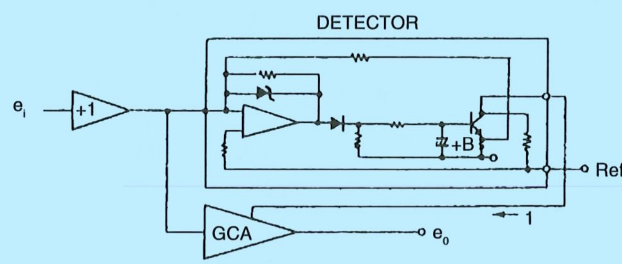


Figure 62 Expander Principle of Operation

PIONEER ELECTRONIC CORPORATION
4-1, Meguro 1-chome, Meguro-ku, Tokyo 153, JAPAN

PIONEER ELECTRONICS SERVICE, INC
P.O. Box 1760, Long Beach, CA. 90801-1760 U.S.A.

PINEER ELECTRONIC (EUROPE) N. V.
Haven 1087, Keetberglaan, 1.,9120 Melsele BELGIUM

Published by Service Engineering Training Section
Service Administration & Technical Information Dept.
Product Reliability & Customers Service Div.